SONY

Digital Reverberator

Operating Instructions

DPS-R7

WARNING

FOR CUSTOMERS IN THE UNITED STATES

Owner's Record

The model and serial numbers are located on the left side. Record these numbers in the spaces provided below. Refer to these numbers whenever you call upon your Sony dealer regarding this product.

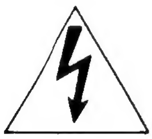
Model No. DPS-R7 Serial No. ____________________

WARNING

To prevent fire or shock hazard, do not expose the unit to rain or moisture.

CAUTION
RISK OF ELECTRIC SHOCK
DO NOT OPEN

CAUTION: TO REDUCE THE RISK OF ELECTRIC SHOCK,
DO NOT REMOVE COVER (OR BACK).
NO USER-SERVICEABLE PARTS INSIDE.
REFER SERVICING TO QUALIFIED SERVICE PERSONNEL.



This symbol is intended to alert the user to the presence of uninsulated "dangerous voltage" within the product's enclosure that may be of sufficient magnitude to constitute a risk of electric shock to persons.

This symbol is intended to alert the user to the presence of important operating and maintenance (servicing) instructions in the literature accompanying the appliance.

For detailed safety precautions, see the leaflet "IMPORTANT SAFEGUARDS".

INFORMATION

This equipment generates and uses radio frequency energy and if not installed and used properly, that is, in strict accordance with the manufacturer's instructions, may cause interference to radio and television reception. It has been type tested and found to comply with the limits for a Class B computing device in accordance with the specifications in Subpart J of Part 15 of FCC Rules, which are designed to provide reasonable protection against such interference in a residential installation. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

Reorient the receiving antenna
Relocate the equipment with respect to the receiver
Move the equipment away from the receiver
Plug the equipment into a different outlet so that equipment and receiver are on different branch circuits.

If necessary, the user should consult the dealer or an experienced radio/television technician for additional suggestions. The user may find the following booklet prepared by the Federal Communications Commission helpful:
"How to Identify and Resolve Radio-TV Interference Problems". This booklet is available from the U.S. Government Printing Office, Washington, DC 20402, Stock No. 004-000-00345-4.

- The shielded interface cable recommended in this manual must be used with this equipment in order to comply with the limits for a computing device pursuant to Subpart J of Part 15 of FCC Rules.

FOR CUSTOMERS IN CANADA

CAUTION:
TO PREVENT ELECTRIC SHOCK, DO NOT USE THIS POLARIZED AC PLUG WITH AN EXTENSION CORD, RECEPTACLE OR OTHER OUTLET UNLESS THE BLADES CAN BE FULLY INSERTED TO PREVENT BLADE EXPOSURE.

This apparatus complies with the Class B limits for radio noise emissions set out in Radio Interference Regulations.

FOR CUSTOMERS IN THE UNITED KINGDOM

WARNING
THIS APPARATUS MUST BE EARTHED

IMPORTANT
The wires in this mains lead are coloured in accordance with the following code:

Green-and-yellow :	Earth
Blue :	Neutral
Brown :	Live

As the colours of the wires in the mains lead of this apparatus may not correspond with the coloured markings identifying the terminals in your plug proceed as follows: The wire which is coloured green-and-yellow must be connected to the terminal in the plug which is marked by the letter E or by the safety earth symbol ⏚ or coloured green or green-and-yellow. The wire which is coloured blue must be connected to the terminal which is marked with the letter N or coloured black. The wire which is coloured brown must be connected to the terminal which is marked with the letter L or coloured red.

Precautions

On Safety
- To avoid electrical shock, do not open the cabinet. Refer servicing to qualified personnel only.
- Before connecting the unit to the power source, check that the operating voltage of your unit is the same as the local power line voltage. The operating voltage is indicated on the nameplate on the left side of the unit.
- Should anything fall into the cabinet, unplug the unit and have it checked by qualified personnel before operating it any further.
- If the unit will not be used for an extended period, unplug it from the wall outlet. To disconnect the cord, pull it out by the plug. Never pull the cord itself.
- The unit is not disconnected from the mains (AC power source) as long as it is connected to the mains outlet, even if the unit itself has been turned off.

On Installation
- Allow adequate air circulation to prevent internal heat build-up.
- Do not place the unit on a surface (rugs, blankets, etc.) or near materials (curtains, draperies, etc.) that may block the ventilation holes.
- Do not install the unit near heat sources such as radiators or air ducts or in a place subject to direct sunlight, excessive dust, mechanical vibration or shock.
- The unit is designed for operation in a horizontal position. Do not install it in an inclined position.
- Keep the unit away from equipment with strong magnets, such as microwave ovens or large loudspeakers.
- Do not place any heavy object on the unit.

On Operation
- When the unit is not in use, turn the power off to conserve energy and to extend its life.

On Cleaning
- Clean the cabinet, panel and controls with a dry soft cloth, or a soft cloth slightly moistened with a mild detergent solution.
- Do not use any type of solvents, such as alcohol or benzine, which might damage the finish.

On Repacking
- Do not throw away the carton and packing materials. They make an ideal container to transport the unit.

If you have any questions about the unit, contact your Sony service facility.

CAUTION!
Danger of explosion if battery is incorrectly replaced. Replace only with the same or equivalent type recommended by the equipment manufacturer. Discard used batteries according to manufacturer's instructions.

Table of Contents

Overview of the DPS-R7

The DPS-R7 is a digital reverberator equipped entirely with Sony's digital and audio technology at its highest level of sophistication which was released before with the Digital Reverberator, DRE-2000 and MU-R201 and achieved much appraisal.

Quality-conscious design – A/D and D/A high performance converter
The DPS-R7 converts the incoming analog signal to a digital signal, passes it through various effects, then re-converts it to an analog singnal before output. The determinant to the sound quality is the conversion mechanism that adopts the 18-bit oversampling stereo A/D converter and the pulse D/A converter of 40.96 MHz. These account for highly accurate, less deteriorated effects.

User-friendly and comfortable operation
The large size backlit LCD of 40 characters by 2 lines makes it possible to proceed with smooth operation while viewing the operating condition in real time. Moreover, the LCD display incorporates an on-line manual (in English) which displays information required for operation.

Abundant preset memory settings
The unit has a hundred variations of the effects created by musicians, sound mixers and acoustic engineers around the world in its preset memory. This will help you select and replay immediately the desired effects for a particular purpose.

Sound creation of any kind
The EDIT function allows you to modify the presets or to create some individual effects. Besides the preset memory for a hundred effects, the unit has a so-called user memory where you can save up to 256 effects you are going to create. Using this memory allows more varicolored play of effects.

Wide range of effects
The DPS-R7 consists mainly of a reverberation block together with an input block, a pre-effect block, a post-effect block and an output block for signal processing.
For Processing signals with stereo-input/stereo-output in the reverberation block, one of 5 types of ST-ST algorithms is available, while the monaural-input/stereo-output processing allows you to take any two types of MONO-ST algorithm. One of the six types of algorithms can be used in the pre-effect block and one of seven in the post-effect block.

By combining these blocks and algorithms used in the blocks, a wide range of effects will be able to be created according to the input source.

Remote control
The remote commander (not supplied) makes it possible to remotely control the unit.

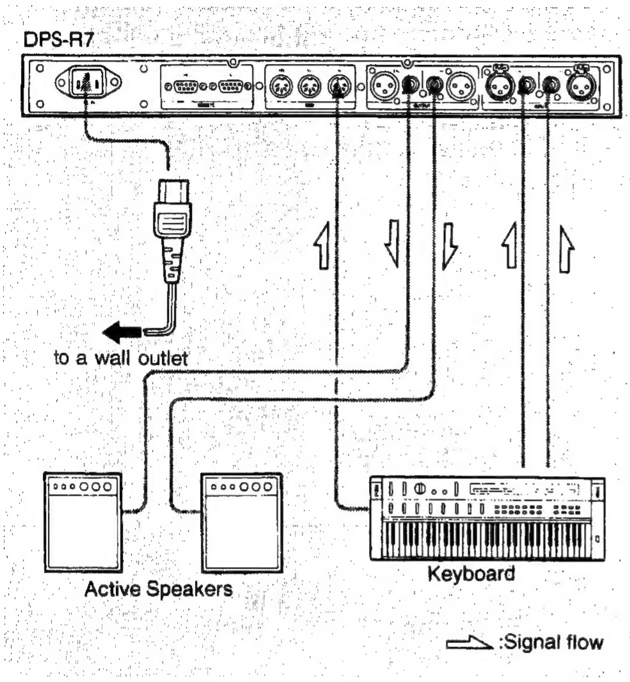
2 types of I/O connectors are provided
The DPS-R7 has XLR connectors (balanced-type) and phone jacks so that it can be connected to instruments, recording equipment or PA (public address) equipment.

Linkage with MIDI equipment
The MIDI device incorporated in the DPS-R7 can receive program change signals from another MIDI equipment connected so that the DPS-R7 can be controlled by the MIDI equipment connected. Thus, it can function as an effector when connected to digital instruments. In addition, controls from PC's or MIDI sequencers are very helpful for composition.

Hooking Up a System

Turn all the power off before making connections, and connect the AC power cord last.

Fundamental Connections as an Effector



1. Connect a keyboard to the INPUT jacks, or the MIDI IN connector.

2. Connect active speakers to the OUTPUT jacks.

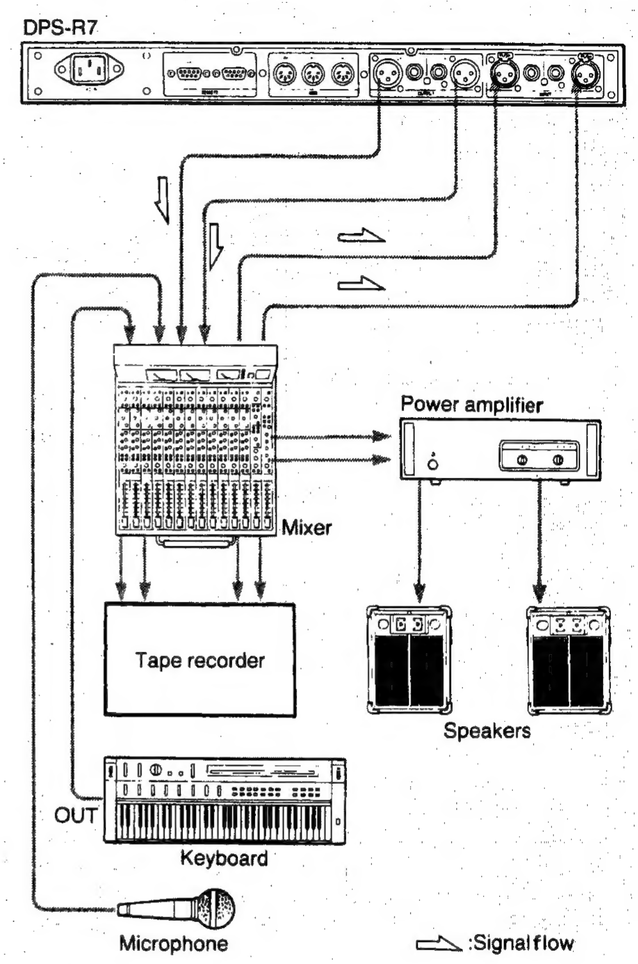
3. Insert the AC power cord firmly into the AC IN jack.

4. Connect the AC power cord to a wall outlet.

For models equipped with a voltage selector
Check to confirm that the voltage selector is set to the local power line voltage. If not, set the selector to the correct position before connecting the AC power cord to a wall outlet.

Notes:
- Be sure to insert the plugs firmly into the jacks. Loose connection may cause hum and noise.
- Leave a little slack in the connecting cord to allow for inadvertent shock or vibration.
- Connect the AC power cord last.
- Connections with some equipment of which the output capacity is very high may result in sound distortion. When this happens, turn the INPUT control to lower the input level, or turn the output level of the equipment connected to the DPS-R7.

Fundamental Connections for Recording

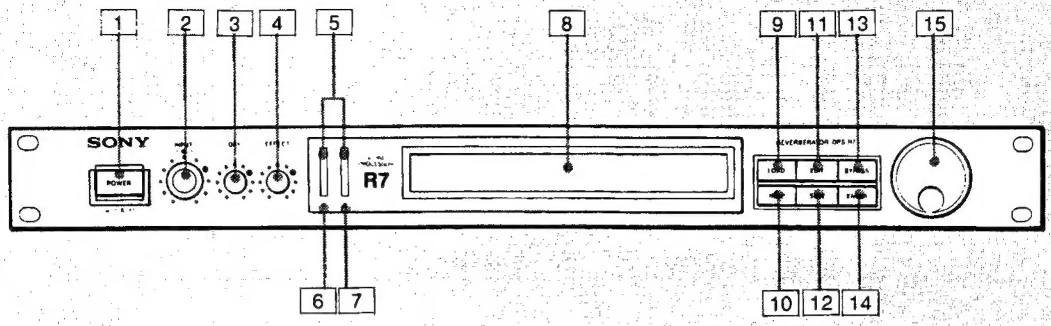


Notes:
- If only one channel is used, connect it to the INPUT CH1 and set the input mode in the System block to "mono".
  See page 40.
  This allows you to obtain the same result as you do when the mode is set to "stereo" and the same signal is input both to the INPUT CH1 and INPUT CH2.
- Be sure to input the signal with the reference level of +4 dB to the XLR-3-31 connector.
- Since the reference level of phone jack is set at −10 dB, any input signal exceeding the maximum input level of +10 dB causes clipping to the amplifier before the input volume and sound may be distorted.
- An optional remote commander RM-DPS7 can be connected to the REMOTE IN connector to remotely control this unit.

Identifying the Parts

To be continued ►

Front panel



1 POWER switch
Turns the unit on and off. When the power is on, the back light of the display window illuminates and the last indication appears. For a few seconds after switching on, the sound being input will be directly output since the bypass function works.

2 INPUT control
Adjusts the input levels of two channels independently. The outside knob controls channel 1 and the inside knob controls channel 2. Since they are linked with each other, hold whichever you do not use to adjust for one channel only.

3 DRY (original sound) control
Adjusts the output level of the source signal which is clear of any effect. Using this control in conjunction with the EFFECT control can adjust the balance between the source signal and the effect signal when mixing them. To output only the effect signal, set the control to "0".

4 EFFECT control
Adjust the output level of the effect signal. Using this control in conjunction with the DRY control can adjust the balance between the source signal and the effect signal when mixing them.

5 Input level meter
Indicates input level for both channels independently. Adjusts the INPUT control so that 0 dB is lit when the signal of the reference level is input. 0 dB means the head room of 20 dB. When the signal exceeds this head room, "OVER" illuminates.
This meter does not work when the BYPASS button is pressed.

6 MIDI indicator
Illuminates when the unit receives the MIDI program change signal or control change signal.

7 REMOTE indicator
Illuminates when the unit receives the signal from an optional remote commander.

8 Display window
A display of 40 characters by 2 lines on which names of called memory, parameter values and/or messages are indicated. Displayed indication is easy to read in a dark hall or a studio due to the back lighting.

9 LOAD button
Press the button to call up the memory.

10 HELP button
Press the button to display information or messages necessary to proceed with operation.

11 EDIT button
Press the button to change parameter values in the memory.

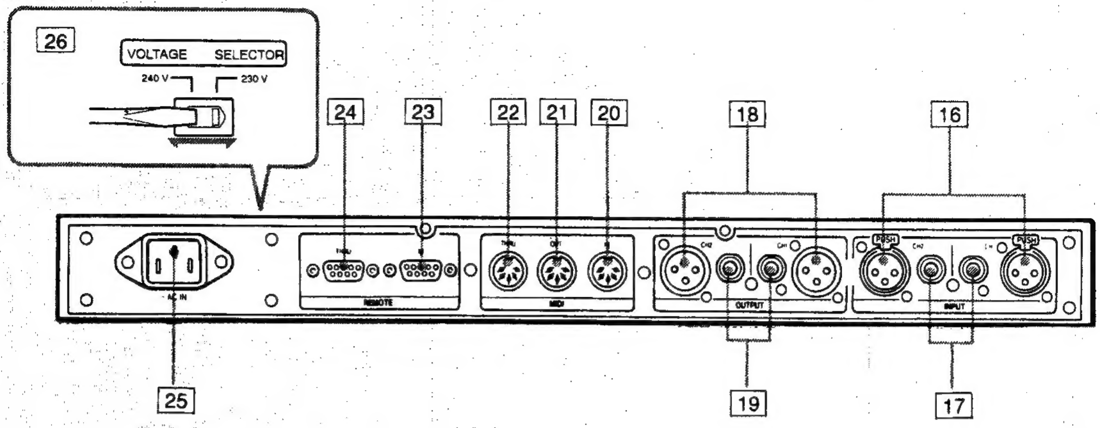
12 SAVE button
Press the button to save the effects individually created in the user memory.

13 BYPASS button
Press the button to output the input signal directly.

14 ENTER button
Press the button every time you have finished selecting or setting parameters.

15 Operating dial
Selects preset numbers and/or sets parameters.

Rear panel



16 **INPUT CH1/CH2 terminal (XLR-3-31 connector)**
Balanced-type terminals for input of ch1 and ch2.

17 **INPUT CH1/CH2 terminal (phone jack)**
Phone jacks for input of ch1 and ch2.

18 **OUTPUT CH1/CH2 terminal (XLR-3-32 connector)**
Balanced-type terminals for output of ch1 and ch2

19 **OUTPUT CH1/CH2 terminal (phone jack)**
Phone jacks for output of ch1 and ch2.

When both XLR connectors and phone jacks are used, the equipment connected to the phone jacks will be given a priority.

20 **MIDI IN terminal (DIN 5-pin)**
Input terminal for the MIDI signal. A commercially available MIDI cable can be connected between this terminal and a MIDI OUT (or THRU) terminal of another MIDI equipment.

21 **MIDI OUT terminal (DIN 5-pin)**
Outputs the MIDI signal generated in this unit.

22 **MIDI THRU terminal (DIN 5-pin)**
This terminal directly outputs the MIDI signal received from the MIDI IN terminal. A commercially available MIDI cable can be connected between this terminal and a MIDI IN terminal of another MIDI equipment.

23 **REMOTE IN terminal (D-Sub 9-pin)**
The terminal to which an optional remote commander is connected. The remote commander enables you to remotely control the unit.

24 **REMOTE THRU terminal (D-Sub 9-pin)**
This terminal directly outputs the signal received from an optional commander through the REMOTE IN terminal. This terminal should be connected to a REMOTE IN terminal of any effector of the DPS series.

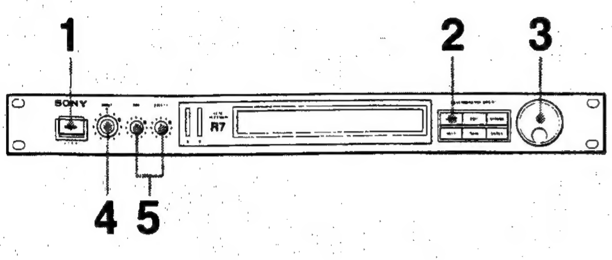
25 **AC IN jack**
Use the supplied power cord to plug in a power outlet.

26 **Voltage selector**
(only for UK and European model)
Set the voltage selector to the correct position before connecting the AC power cord to a wall outlet.

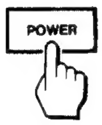
Let's Try to Operate Your DPS-R7

The DPS-R7 has a hundred effects stored in the preset memory. Now first, listen to these effects one by one, referring to "Hooking Up a System" (page 6) and "Preset Memory List" (separate volume).

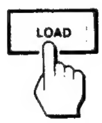
Selecting a Preset Effect

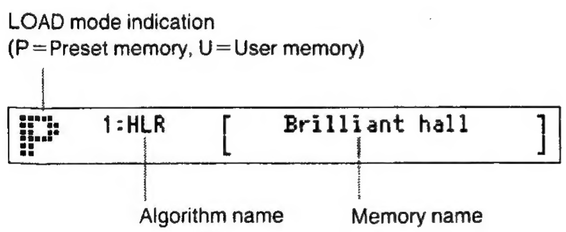


1. Turn on the power.

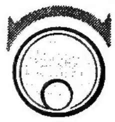


2. Press the LOAD button.





3. Turn the dial to select a desired preset number (P1 to P100*, U1 to U256**)

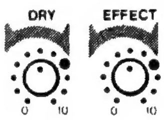


* For the contents of the preset memory, refer to the "Preset Memory List" (separate volume).

** User memory numbers (U1 to U256) become available only after presetting for the numbers are completed.

4. Turn the INPUT control to adjust the input level.

5. Turn the DRY and EFFECT controls to adjust the balance between source signals and effect signals to obtain a desired sound.



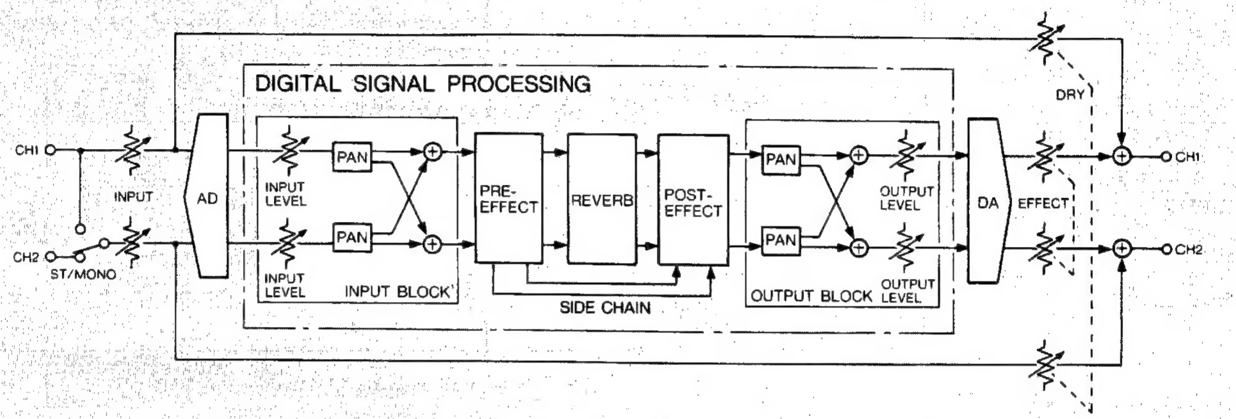
Before turning on the power of connected equipment
Turn down the volume of each equipment to prevent unexpectedly high sound from occurring.

To output the input signals directly
Press the BYPASS button. The input signals will be output directly. To release the bypass function, press the button again.

Overview of the Signal Processing Blocks

The digital signal converted by the A/D converter is then processed through Input, Pre-effect, Reverberation, Post-effect and Output blocks sequentially.
Then it comes into the D/A converter.

General Block Diagram



Note
When using the bypass function, signals input to ch1/ch2 bypasses the electric circuit and are directly output to the output terminals. When switching off the unit, the system automatically applies the bypass function.

Since the digital signal processing has a 12 dB margin against the full-bit output signal from the A/D converter, the signal level raised within 12 dB by EQ (equalizer) parameters in the digital signal processing blocks can be regulated by simply changing the output level to prevent the clipping.
If the signal is supposed to be raised more than 12 dB in the digital processing blocks, lower the input level.

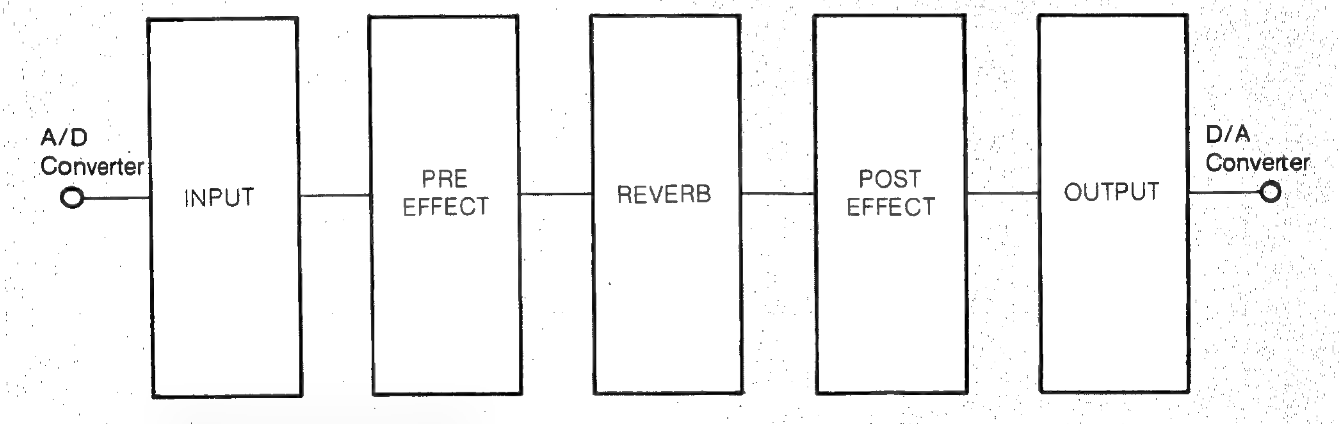
The analog signal processing has a gain of 10 dB for each input and output. Turn the INPUT, DRY and EFFECT controls to make the I/O level suitabale for the equipment connected to the DPS-R7. Setting the control to the two o'clock position (largest point on the scale) produces a gain of 0 dB approximately.

Algorithm "OFF"
The blocks such as Pre-effect, Post-effect and Reverberation contain the algoritm called "OFF". When turning "OFF" the block, that is, selecting OFF algorithm in the block, signal can bypass the block.
To turn "OFF" the block in a user memory setting, load the corresponding block, by using B. LOAD function, from a preset memory setting in which the algorithm "OFF" is selected.
By turning "OFF" a block, you can save the data area in the user memoy.

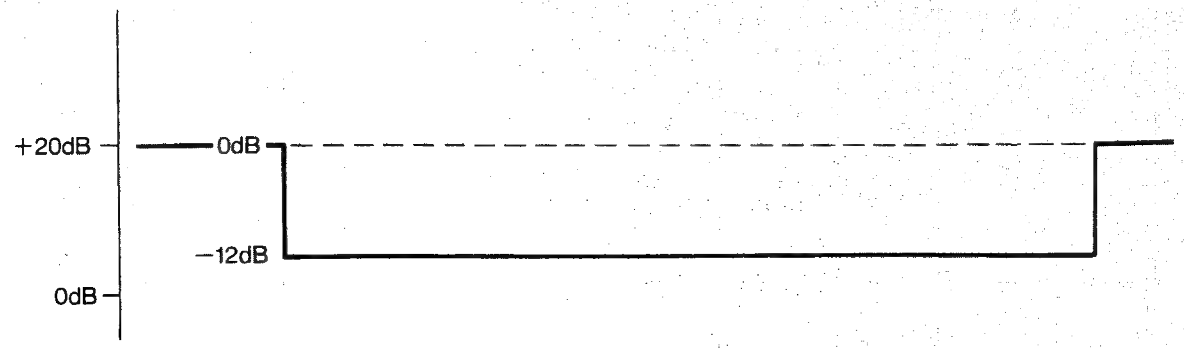
What can be done with each block?

Input block	— the first block to receive the digital signals. See page 12 for details.
Pre-effect block	— pre-processing block closely related to the Post effect block. See pages 13 to 20.
Reverberation block	— the most important block in creating effects.See pages 21 to 32.
Post-effect block	— post-processing block closely related to the Pre-effect block. See pages 33 to 37.
Output block	— the final block to send signals to D/A converter. See page 38

Signal flow diagram



Analog level-digital level diagram



Input Block

This block receives the signals from the A/D converter to give them level, phase and panpot processing.

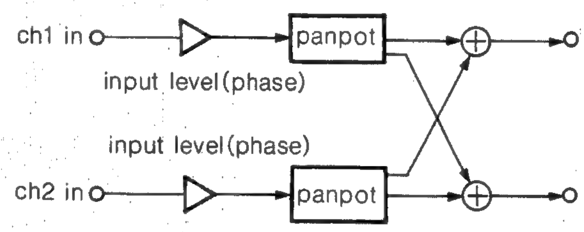
Abbreviations of parameter names

The parameter name shown in this manual may be abbreviated from time to time as shown below.

auto panner → autopan
early reflection → early ref., e.ref.
reverberation → reverb
rotate → RT
predelay → PD

Parameters	MIN and MAX
input level (ch1, ch2)	0 – 100%
input phase (ch1, ch2)	normal/inverse
panpot (ch1, ch2)	0 – 100%
panpot limit min (ch1, ch2)	0 – 100%
panpot limit max (ch1, ch2)	0 – 100%

Note:
If the panpot is set to 0%, signals pass through the block and if it is set to 100%, input signals for ch1 is output to ch2 and vice versa. Available values for panpot lie between limit min and max.



Pre-effect Block

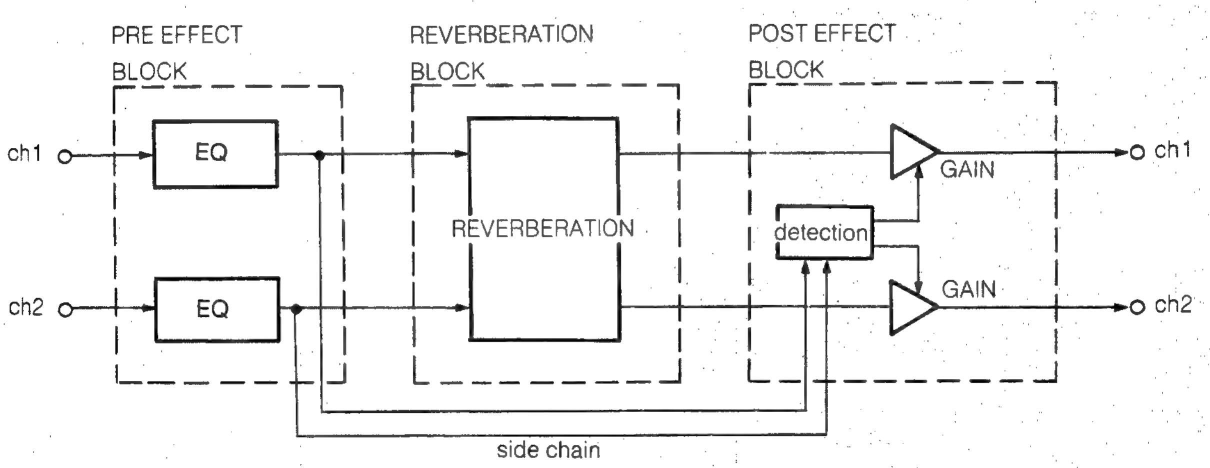
To be continued ►

This block processes the incoming signals from the Input block then outputs them to the Reverberation block. Signal processing in this block uses 6 types of specific algorithms for the preset memories.

When editing a preset memory, examine which algorithm is used in the preset memory. Parameters also varies as the algorithm changes.

What is the side chain?

The side chain is a line by which signals are directly output from the Pre-effect block to the Post-effect block. This is effective when you have chosen the gate (Algorithm 6 GTE) or the auto panner (Algorithm 7 APN) in the Post-effect block. In other words, as shown in the diagram below, By bypassing the Reverberation block (that is, By sending the signals directly from the Pre-effect block to the Post-effect block), a well responsed gate effect can be obtained without any signal processing in the Reverberation block. (With the gate incorporated in the DPS-R7, you can select input signals for the gate processing. Therefore, signals output from the Reverberation block can be also selected as an input source to the gate processing.)

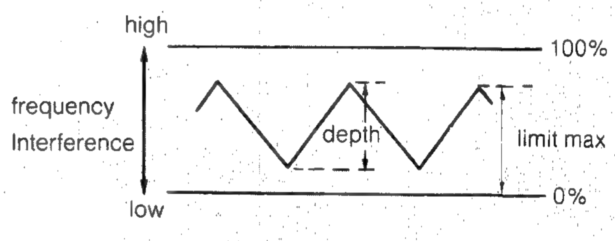


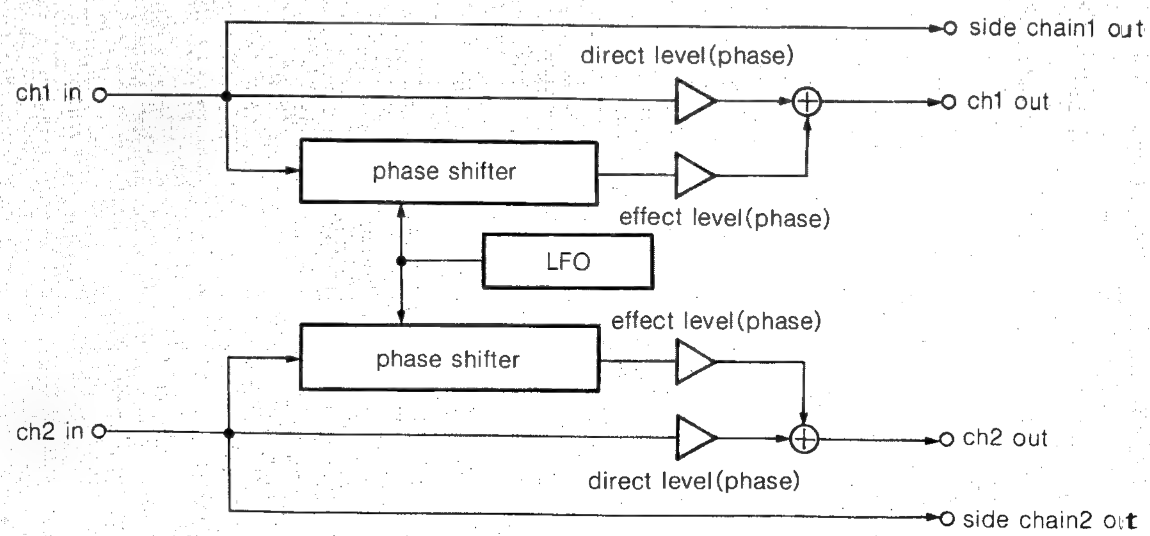
Algorithm 1 — Phase Shifter PHS

This is a dual channel phase shifter. One LFO (Low Frequency Oscillator) which determines the amount of a phase shift outputs signals of the same phase to each channel.
The effect is created by the phase interference and may sound like sparkling. Raising the resonance may give the tone loud manner. Also changing the mixing ratio of the direct sound and the effect sound, or shifting the phases of resonance brings variation in tones. So listen to these changes as you set the parameters.

Parameters	MIN and MAX
phase shifter on/off	on/off
LFO freq	0.1 – 20Hz
depth depth : limit max	0 – 100% 0 – 100%
resonance level (ch1, ch2) resonance phase (ch1, ch2)	0 – 99.9% normal/inverse
direct level (ch1, ch2) direct phase (ch1, ch2)	0 – 100% normal/inverse
effect level (ch1, ch2) effect phase (ch1, ch2)	0 – 100% normal/inverse

LFO Depth
LFO depth is a parameter which defines an amplitude of the LFO waveform and its offset. The more the LFO waveform comes up to 100%, the higher the frequency band where interference occurs becomes.





To be continued ►

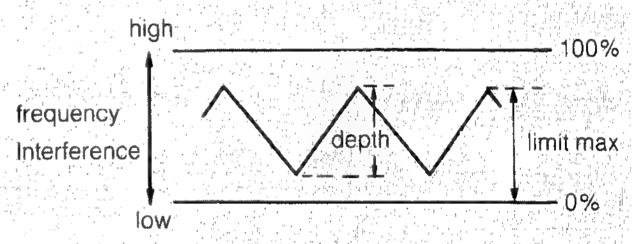
Algorithm 2 Flanger FLG

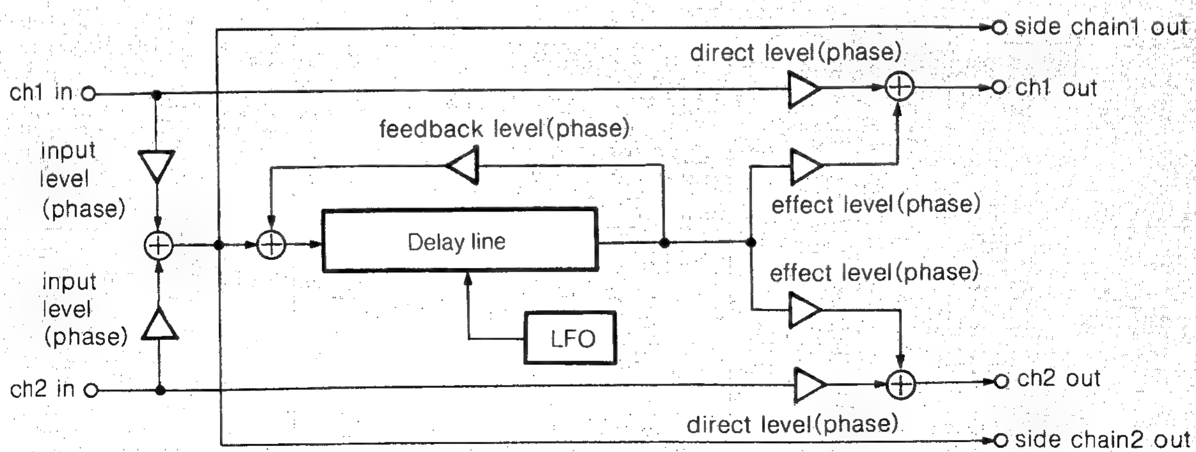
English Pre-effect Block

The signals of ch1 and ch2 are mixed together for input to one unit of flanger. Phase can be changed in input, output and feedback signals, enabling a flanger effect to spread. The effect is different from that of the Phase shifter (Algorithm 1) and sounds like rubbing, which is characterized by a comb filter effect by waveform interference.
Properly raising the feedback gives the tone a manner. If you apply this effect to the sounds containing a lot of overtones like cymbals, you can create sounds which have intervals as well as a feeling of swelling.

Parameters	MIN and MAX
flanger on/off	on/off
input level (ch1,ch2) input phase (ch1,ch2)	0 – 100% normal/inverse
LFO freq	0.1 – 20Hz
depth depth : limit max	0 – 100% 0 – 100%
feedback level feedback phase	0 – 99.9% normal/inverse
direct level (ch1, ch2) effect phase (ch1, ch2)	0 – 100% normal/inverse
effect level (ch1, ch2) effect phase (ch1, ch2)	0 – 100% normal/inverse

LFO Depth
LFO depth is a parameter which defines an amplitude of the LOF waveform and its offset. The more the LFO waveform comes up to 100%, the higher the frequency band where interference occurs becomes.

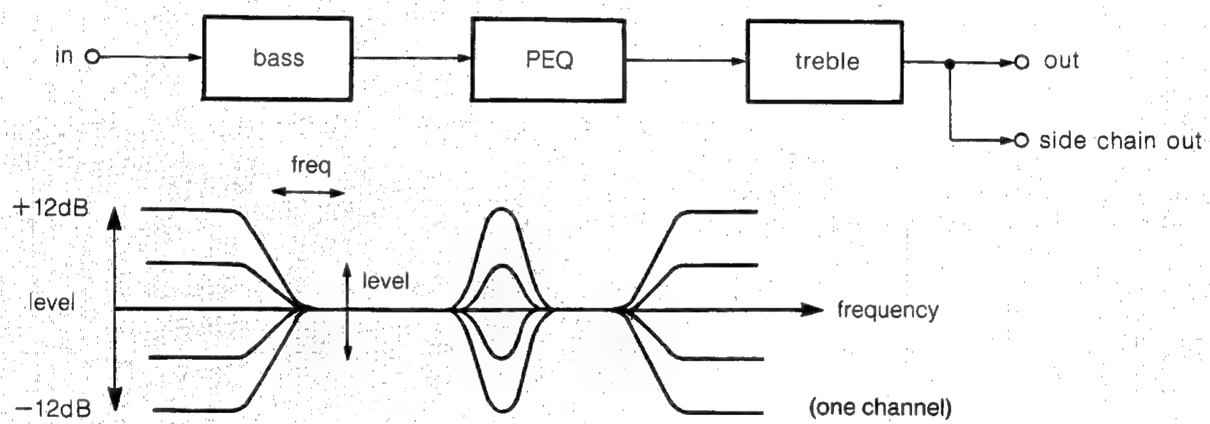




Algorithm 3 Stereo Equalizer SEQ

This is a dual channel digital equalizer and consists of 3 band equalization (bass, treble and peaking) independently operated for ch1 and ch2.

Parameters	MIN and MAX
stereo EQ on/off	on/off
bass freq (ch1, ch2) bass level (ch1, ch2)	16 – 6.3kHz −12 – +12dB
treble freq (ch1, ch2) treble level (ch1, ch2)	400 – 18.0kHz −12 – +12dB
PEQ freq (ch1, ch2) PEQ level (ch1, ch2) PEQ q (ch1, ch2)	63 – 18.0kHz −12 – +12dB 0.267/0.667/1.414/2.145 4.319/8.651/17.31/34.62

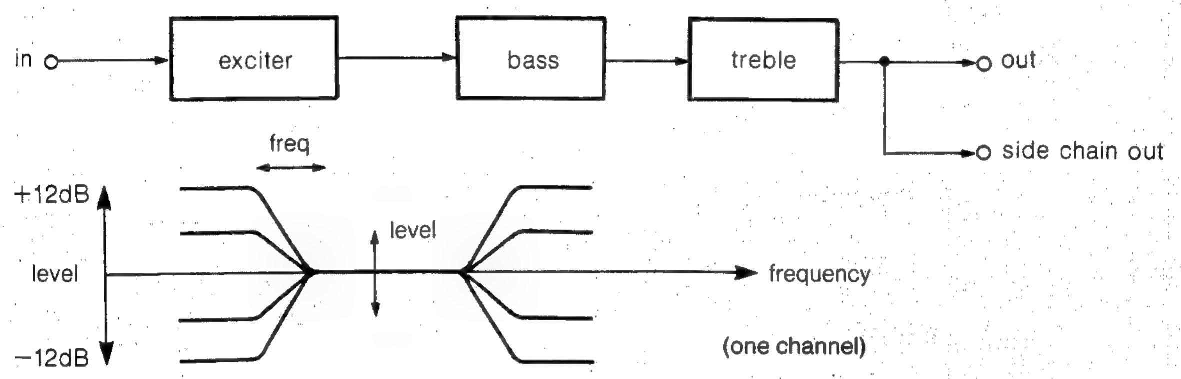


To be continued ►

Algorithm 4 Stereo Exciter + Stereo EQ (Equalizer) SXE

This is a dual channel digital equalizer with a dual channel exciter.
The equalizer consists of a 2 band equalization (bass and treble) independently operated for the channels.
The exciter clarifies the shape of incoming signals and modulates the signals to create a stressed sound.

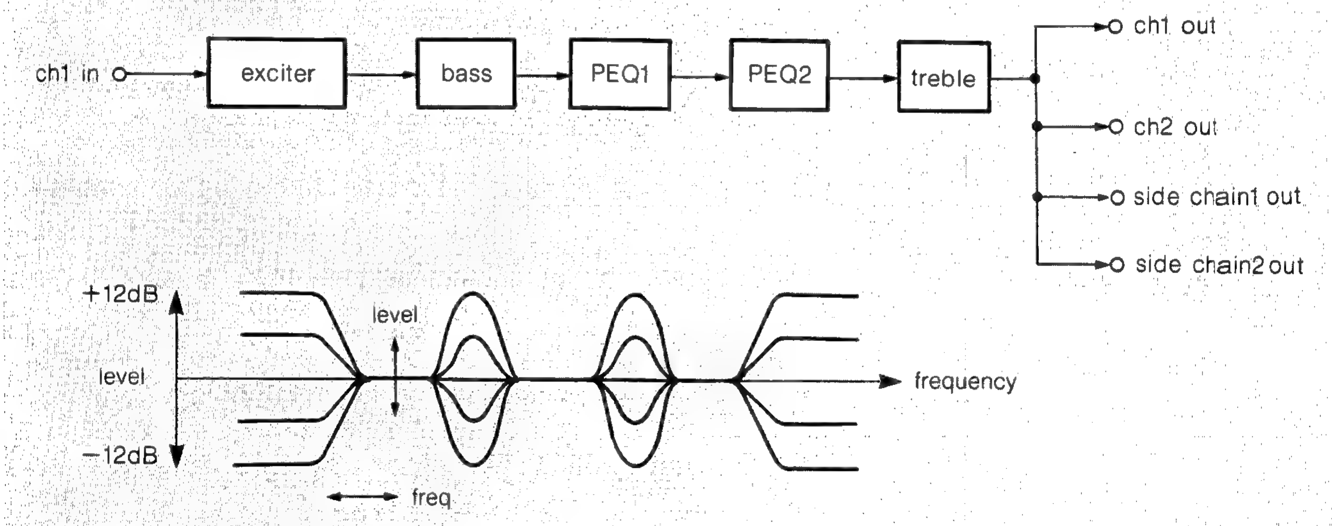
Parameters	MIN and MAX
stereo exciter + EQ on/off	on/off
exciter level (ch1, ch2)	−100 – +100%
bass freq (ch1, ch2) bass level (ch1, ch2)	16 – 6.3kHz −12 – +12dB
treble freq (ch1, ch2) treble level (ch1, ch2)	400 – 18.0kHz −12 – +12dB



Algorithm 5 Monaural Exciter + Monaural EQ (Equalizer) MXE

This is a monaural digital equalizer with an exciter. The equalizer consists of 4 band equalization (bass, treble, peaking 1 and peaking 2). No signals can be input from ch2.

Parameters	MIN and MAX
mono exciter + EQ on/off	on/off
exciter level	−100 – +100%
bass freq bass level	16 – 6.3kHz −12 – +12dB
treble freq treble level	400 – 18.0kHz −12 – +12dB
PEQ1 freq PEQ1 level PEQ1 q	63 – 18.0kHz −12 – +12dB 0.267/0.667/1.414/2.145 4.319/8.651/17.31/34.62
PEQ2 freq PEQ2 level PEQ2 q	63 – 18.0kHz −12 – +12dB 0.267/0.667/1.414/2.145 4.319/8.651/17.31/34.62

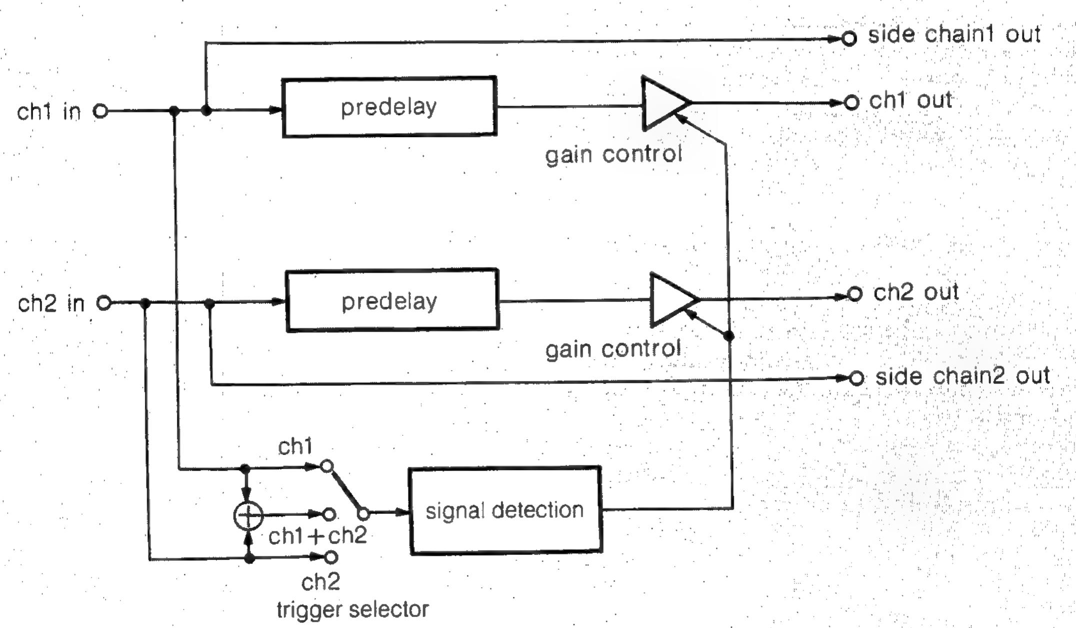


To be continued ►

Algorithm 6 Gate GTE

This is a dual channel gate. The function of a gate is to turn on and off the output signals according to the level of input signals. Making good use of this function, you can set the parameters so as to output only attacking sound or to reduce noise when there is no signal.
Also, you can change the time constant by which the gate is turned on or off. This allows you to change the envelope during attacking or releasing and thus to obtain an effective sound.
In particular, using this in conjucntion with the reverberation for the percussive input signals such as drums, you will be able to obtain a real gated reverberation.
(When using this algorithm as a gated reverberation, also use the same algorithm in the post-effect block.
The predelays on the main lines adjust the time required for signal detection. For example you can apply the predelay to the original sound to create an effect as if it had been passed through the gate processing.

Parameters	MIN and MAX
gate on/off	on/off
trigger select	ch1/ch2/ch1+ch2
attack time	0 – 500msec
release time	0 – 5000msec
threshold level	0 – 100%
hysterisis level	0 – 100%
predelay time (ch1, ch2)	0 – 10words

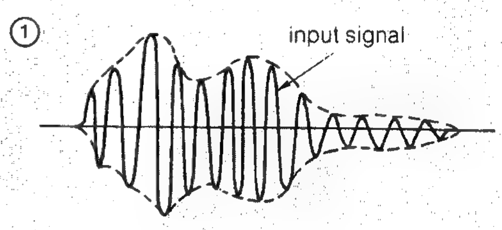


< Signal detecting section >

The signal detecting section detects the level of the incoming signals to control the gate level on the main line. The following describes the activities of this section.

① Envelope detection

The incoming signals selected at the trigger selecting section are detected for envelope waveform at the envelope follower. (See the dotted line in the figure right.)



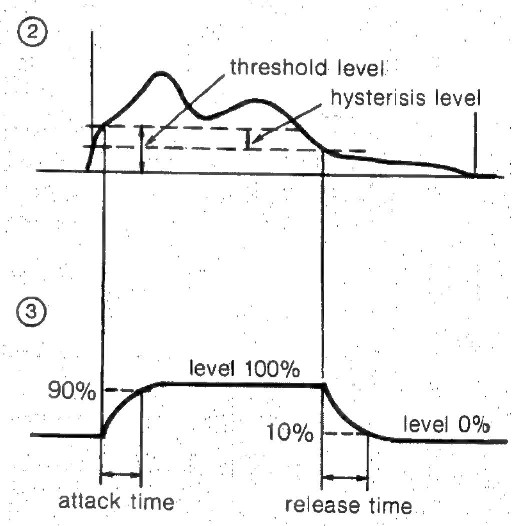
② Threshold Setting

Next, a threshold is set to gain the gate signals from the envelope waveform. With the hysterisis level of 0%, the gate is on if the envelope level exceeds the threshold. On the contrary, it turns off if the envelope level is below the threshold. Since some envelope waveforms may prevent the desired gate output, the signal detecting section allows you to set a hysterisis so that the thresholds for gate-on and gate-off can be set to different values.

Note:
The hysterisis can be set within the range of threshold level.

③ Ouput of gate level control signal

Now the gate level control signal is created from the gate-on/off trigger obtained in the procedure above. The time during which the gate level increases up to 90% of the maximum value from the gate-on trigger is called attack time. On the other hand, the time during which the gate level decreases down to 10% of the maximum value from the gate-off trigger is called release time.



Reverberation Block

This block can be divided into four sub-blocks, REVC, REVS, REV1 and REV2.
These four blocks process signals received from the Pre-effect block and then send them to the Post-effect block. According to the algorithm type used, the signals from the Pre-effect block are processed through different sub-blocks.

When editing a preset memory, examine which algorithm is used in the preset memory.
Parameters also vary as the algorithm type changes.

Stereo-in/stereo-out (ST-ST) type and monaural-in/stereo-out (MONO-ST) type

Algorithm used in the Reverberation block can be characterized into two types; stereo-in/stereo-out (ST-ST) type and monaural-in/stereo-out (MONO-ST) type.

The ST-ST type is used for a stereo-processing of signals of ch1 and ch2 while the MONO-ST type gives different effects to signals of ch1 and ch2. Since these two types have their own signal processing configuration, signals pass through different sub-blocks according to the algorithm type used.

Type of Algorithm	Sub-block to be able to edit
ST-ST	REVC(Reverberation common) sub-block REVS(Reverberation stereo) sub-block
MONO-ST	REVC sub-block REV1 (Reverberation unit1) sub-block REV2 (Reverberation unit2) sub-block

Concerning Parameters

Algorithms contains several parameters. After being familiar with them, you will be able to create any effect exactly as you imagine.

Note:
When editing the Reverberation block, keep the balance of direct sound (DRY control) and effect sound (EFFECT control). Unbalanced adjustment may cause insufficiency in the resulting effect.

Reverberation Parameters

Predelay
Time lag between the direct sound and the reverberation (primary reverberating sound) and the volume (level) of the reverberating sound are user adjustable. Normally the delay time is set to "sync".

2nd predelay
Time lag between the direct sound and slightly more coarse reverberation than the primary reverberation and its volume are user adjustable. Combining this with the predelay parameter will give nuances fine of reverberation.

Cross predelay
Predelay to be output from the predelay box of the opposite channel.
This should be sufficiently delayed more than the predelay.

2nd cross predelay
2nd predelay to be output from the predelay box of the opposite channel.

Early reflection
Initial reflecting sound to be output from the predelay box.

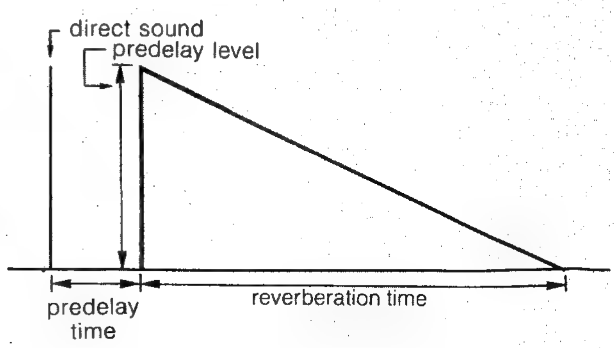
2nd early reflection
Initial reflecting sound with brief reverberation.

Cross early reflection
Initial reflecting sound to be output from the predelay box of the opposite channel.

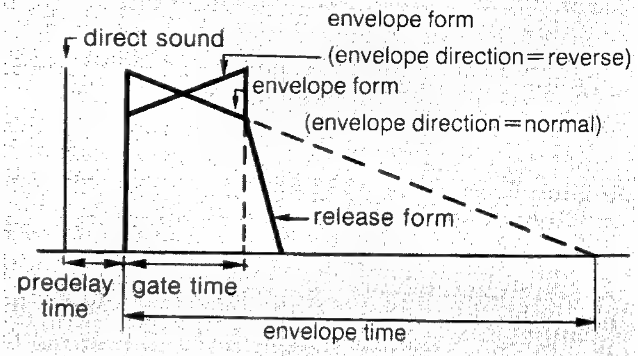
Rotate high, rotate bass, rotate treble
Filters used to change the reverberation time specific to the frequency band. Raising the level too high may cause oscillation.

Spread
Represents the feeling of spreading reverberation. The larger the value becomes, the more reverberation feeling spreads.
Normally the value is set to the same as that of the size parameter.

Size
Represents the time scale of the box which is the most meaningful in each algorithm.
This parameter controls the size of sound field without changing features of the effect.



Gate Reverberation Parameters



Gate time

A parameter in gate reverberation. You can adjust the time from the start of reverberation to the start of its release.

Envelope form

Envelope during the gate time.

Envelope direction

The envelope form, which normally attenuates as time passes, can be reversed when setting the envelope direction parameter to "reverse".
(See the figure to the right)

Envelope time

Used to change the attenuating amount of the envelope.

It is adjusted according to the time during which the amount changes from its peak to −60 dB, provided the gate time is defined as infinite.

Release form

Used to change the release time. It also changes the timbre during the gate time.

Reverberation Block (1) — REVC Block

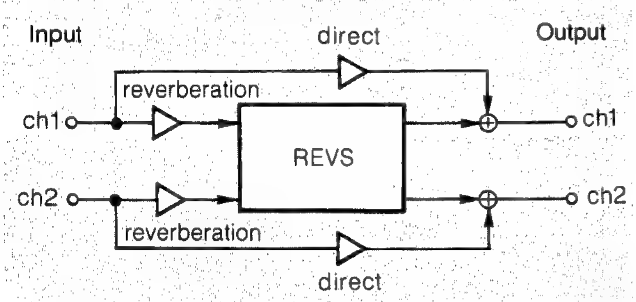
This block is used comonly with both the ST-ST type and the MONO-ST type algorithm.

List of REVC sub-block parameters for ST-ST

Parameters	MIN and MAX
reverb level(ch1, ch2) reverb phase(ch1, ch2)	0 ~ 100% normal/inverse
direct level(ch1, ch2) direct phase(ch1, ch2)	0 ~ 100% normal/inverse

The display shows the blocks to be able to edit.

```
EDIT      ▶REVC     REVS
[ 1/11]    PRE-EF    POST-EF    INPUT  →
```



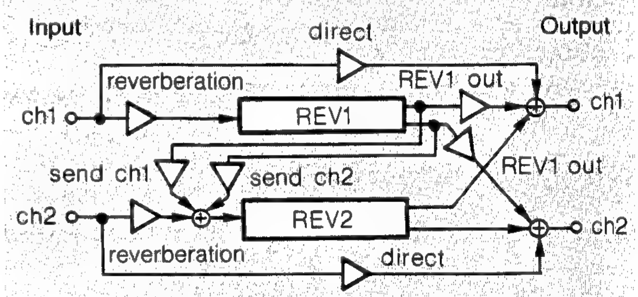
REVC refers to other than REVS.

List of REVC sub-block parameters for MONO-ST

Parameters	MIN and MAX
reverb level(ch1, ch2) reverb phase(ch1, ch2)	0 – 100% normal/inverse
direct level(ch1, ch2) direct phase(ch1, ch2)	0 – 100% normal/inverse
REV1 output level(ch1, ch2) REV1 output phase(ch1, ch2)	0 – 100% normal/inverse
send level(ch1, ch2) send phase(ch1, ch2)	0 – 100% normal/inverse

The display shows the blocks to be able to edit.

```
EDIT      ▶REVC     REV1      REV2
[ 1/12]    PRE-EF    POST-EF   INPUT  →
```



REVC refers to other than REV1 and REV2.

English

Reverberation Block/Reverberation Block (1) – REVC Block

Reverberation Block (2) — REVS Block

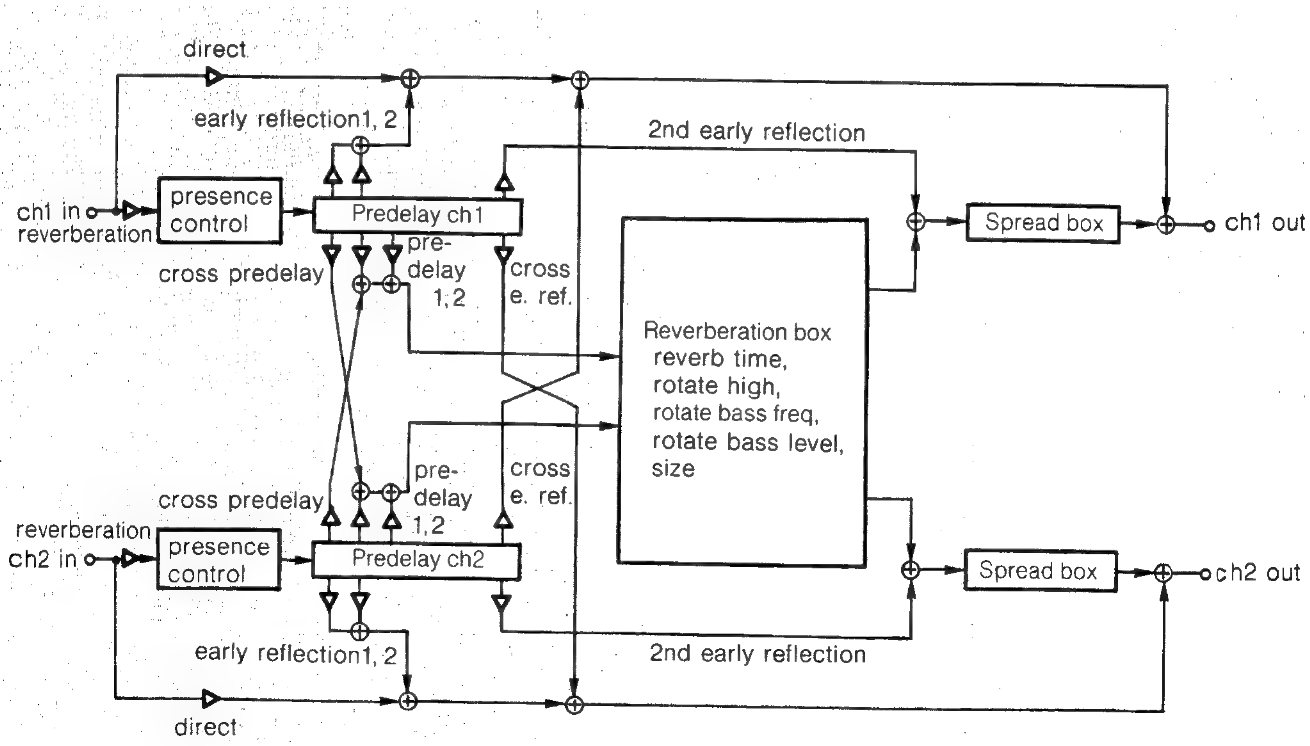
This block is used only with ST-ST type algorithm.

Algorithm 1 — Hall Reverberation (ST-ST) HLR

This is intended for the stereo source. The resulting effect utilizes the localization of the source. This algorithm is suited for creating reverberation in a concert hall.

Parameters	MIN and MAX
reverb time	0.3 – 99.0sec
predelay1,2 time(ch1, ch2) predelay1,2 level(ch1, ch2) predelay1,2 phase(ch1, ch2)	1 – 32767words 0 – 100% normal/inverse
cross predelay time(ch1, ch2) cross predelay level(ch1, ch2) cross predelay phase(ch1, ch2)	1 – 32767words 0 – 100% normal/inverse
early ref. 1,2 time(ch1, ch2) early ref. 1,2 level(ch1, ch2) early ref. 1,2 phase(ch1, ch2)	1 – 32767words 0 – 100% normal/inverse
2nd early ref. time(ch1, ch2) 2nd early ref. level(ch1, ch2) 2nd early ref. phase(ch1, ch2)	1 – 32767words 0 – 100% normal/inverse

Parameters	MIN and MAX
cross early ref. time(ch1, ch2) cross early ref. level(ch1, ch2) cross early ref. phase(ch1, ch2)	1 – 32767words 0 – 100% normal/inverse
presence control	0.003 – 1.000
rotate high rotate bass freq rotate bass level	0.003 – 1.000 25Hz – 6.3kHz −12 – +6dB
spread	0.5 – 1.5
size	0.5 – 1.5



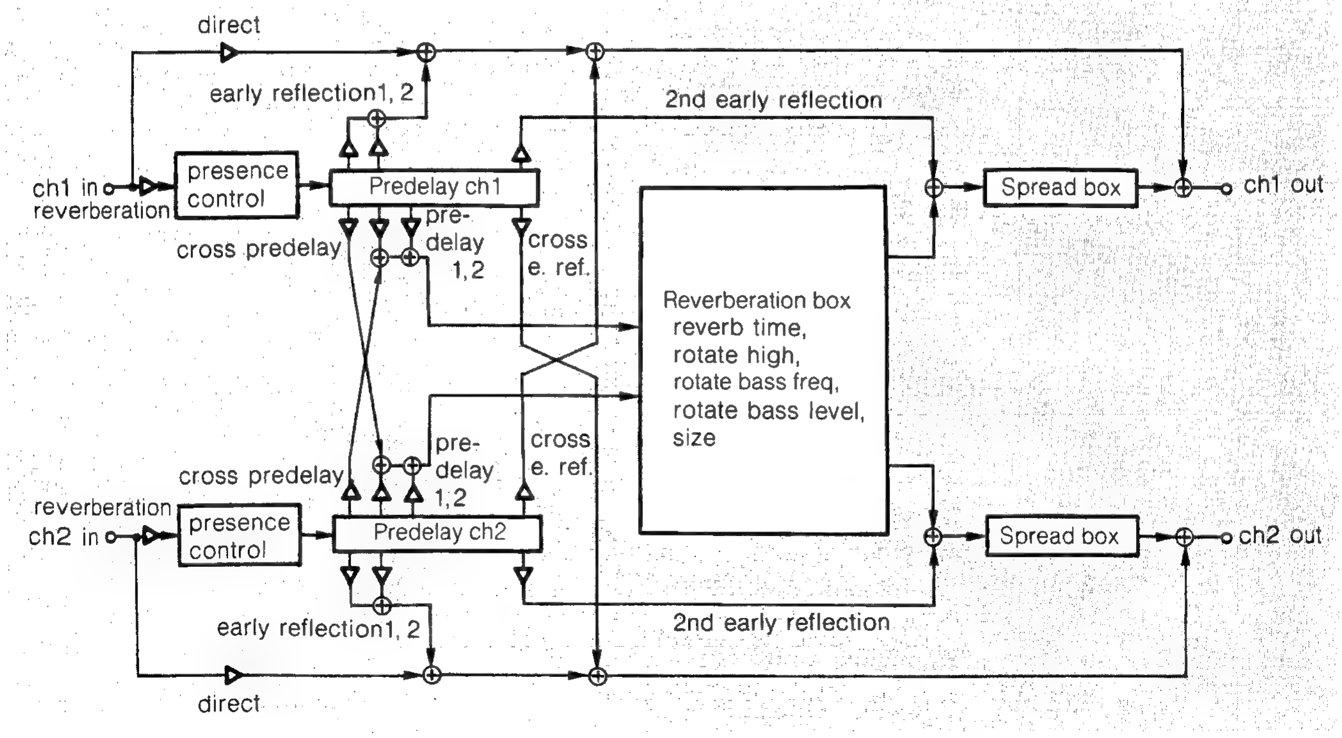
To be continued ►

Algorithm 2 Room Reverberation (ST-ST) RMR

This is intended for the stereo source. The resulting effect utilizes the localization of the source. This algorithm is suited for creating reverberation in relatively small spaces such as a studio, a room in a house, etc.

Parameters	MIN and MAX
reverb time	0.12 – 39.60sec
predelay1,2 time (ch1, ch2) predelay1,2 level (ch1, ch2) predelay1,2 phase (ch1, ch2)	1 – 32767words 0 – 100% normal/inverse
cross predelay time (ch1, ch2) cross predelay level (ch1, ch2) cross predelay phase (ch1, ch2)	1 – 32767words 0 – 100% normal/inverse
early ref. 1,2 time (ch1, ch2) early ref. 1,2 level (ch1, ch2) early ref. 1,2 phase (ch1, ch2)	1 – 32767words 0 – 100% normal/inverse
2nd early ref. time (ch1, ch2) 2nd early ref. level (ch1, ch2) 2nd early ref. phase (ch1, ch2)	1 – 32767words 0 – 100% normal/inverse

Parameters	MIN and MAX
cross early ref. time (ch1, ch2) cross early ref. level (ch1, ch2) cross early ref. phase (ch1, ch2)	1 – 32767words 0 – 100% normal/inverse
presence control	0.003 – 1.000
rotate high rotate bass freq rotate bass level	0.003 – 1.000 25Hz – 6.3kHz −12 – +6dB
spread	0.5 – 2.5
size	0.5 – 1.5

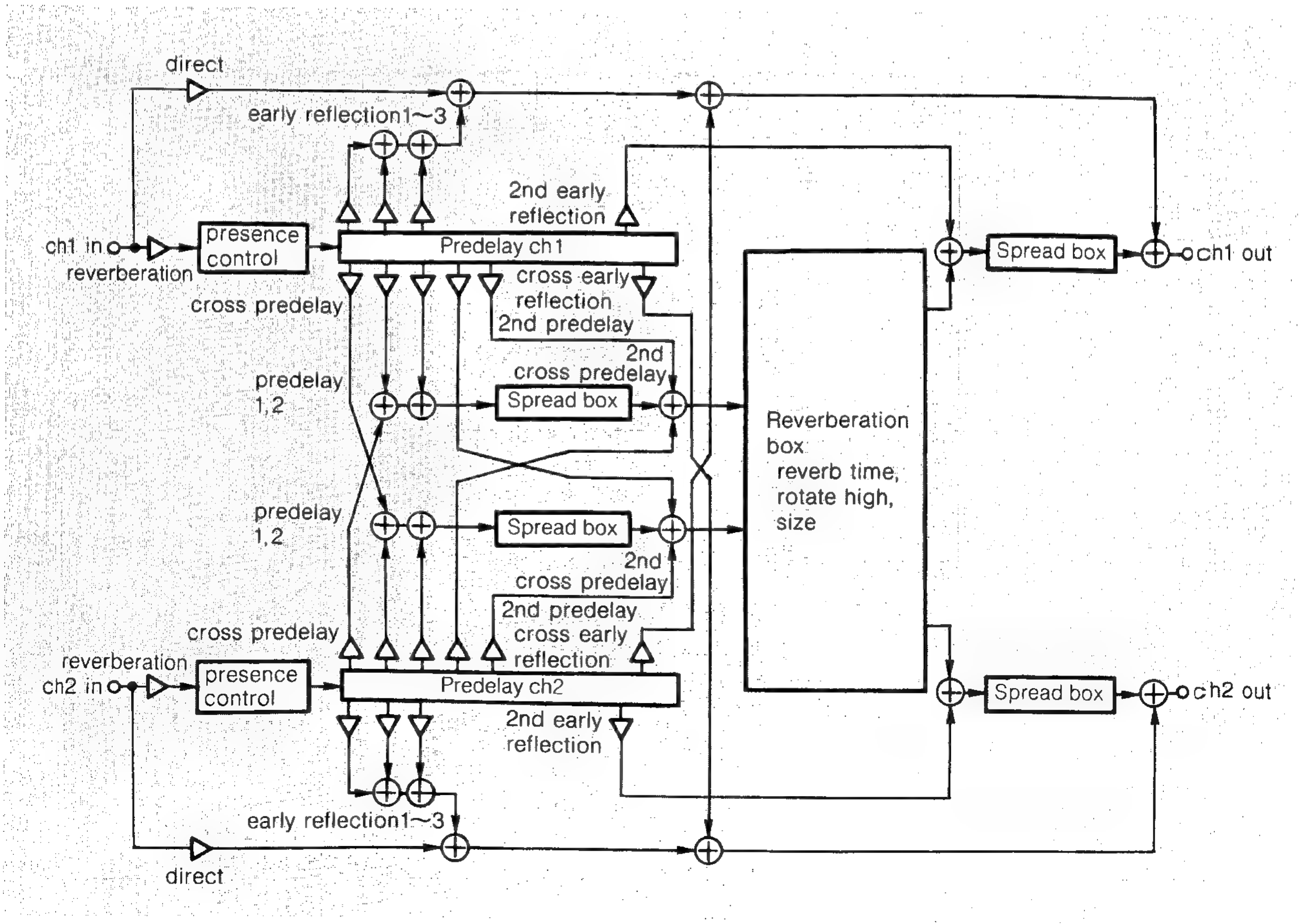


Algorithm 3 Plate Reverberation (ST-ST) PLR

This is intended for the stereo source. The resulting effect has higher reverberation density.
Normally predelay 1 is used as the main reverberation and predelay 2 is secondary.

Parameters	MIN and MAX
reverb time	0.3 – 99.0sec
predelay1,2 time (ch1, ch2) predelay1,2 level (ch1, ch2) predelay1,2 phase (ch1, ch2)	1 – 22527words 0 – 100% normal/inverse
2nd predelay time (ch1, ch2) 2nd predelay level (ch1, ch2) 2nd predelay phase (ch1, ch2)	1 – 22527words 0 – 100% normal/inverse
cross predelay time (ch1, ch2) cross predelay level (ch1, ch2) cross predelay phase (ch1, ch2)	1 – 22527words 0 – 100% normal/inverse
2nd cross predelay time (ch1, ch2) 2nd cross predelay level (ch1, ch2) 2nd cross predelay phase (ch1, ch2)	1 – 22527words 0 – 100% normal/inverse

Parameters	MIN and MAX
early ref. 1 – 3 time (ch1, ch2) early ref. 1 – 3 level (ch1, ch2) early ref. 1 – 3 phase (ch1, ch2)	1 – 22527words 0 – 100% normal/inverse
2nd early ref. time (ch1, ch2) 2nd early ref. level (ch1, ch2) 2nd early ref. phase (ch1, ch2)	1 – 22527words 0 – 100% normal/inverse
cross early ref. time (ch1, ch2) cross early ref. level (ch1, ch2) cross early ref. phase (ch1, ch2)	1 – 22527words 0 – 100% normal/inverse
presence control	0.003 – 1.000
rotate high	0.003 – 1.000
spread	0.5 – 1.5
size	0.5 – 1.5



To be continued ▶

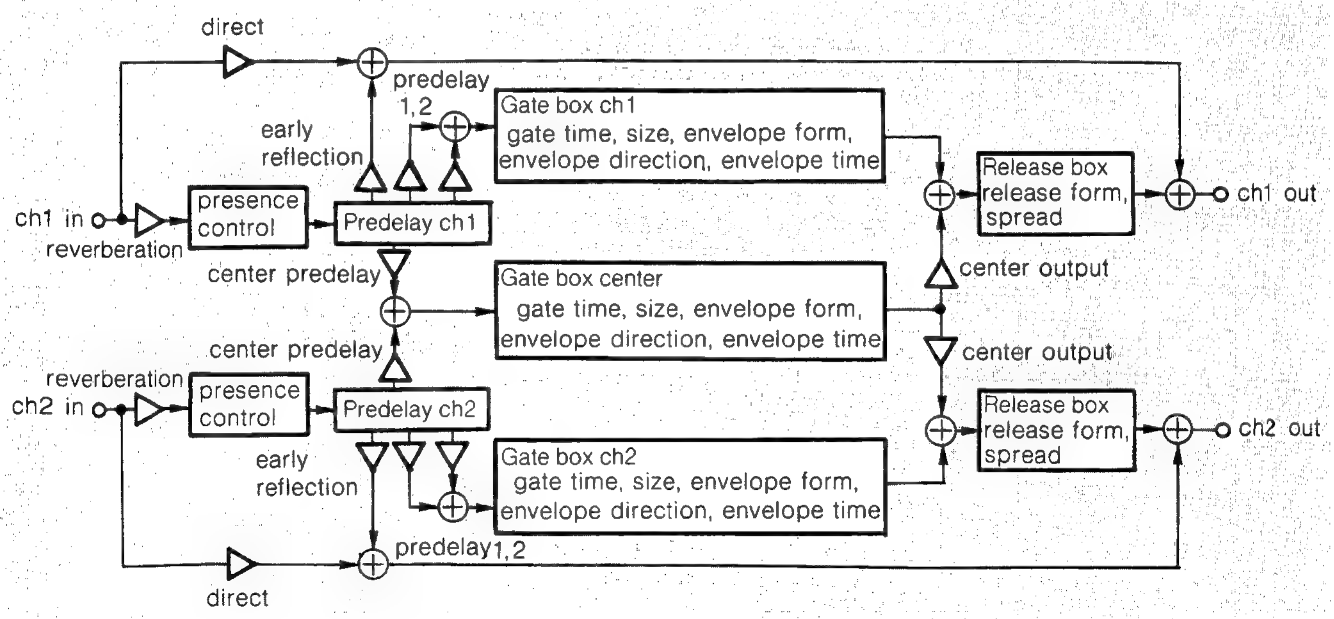
Algorithm 4 Gate Reverberation (ST-ST) GTR

Gate reverberation enables you to assign parameters such as gate time to each box of ch1, ch2 and center.

Parameters	MIN and MAX
gate time (ch1, ch2, center)	1 – 16383words
envelope form (ch1, ch2, center) envelope direction (ch1, ch2, center) envelope time (ch1, ch2, center)	linear1, linear2 exponential1 exponential2 normal/reverse 0.01 – 99.99sec
center output level (ch1, ch2) center output phase (ch1, ch2)	0 – 100% normal/inverse
release from	5 – 100
predelay1,2 time (ch1, ch2) predelay1,2 level (ch1, ch2) predelay1,2 phase (ch1, ch2)	1 – 30719words 0 – 100% normal/inverse

Parameters	MIN and MAX
center predelay time (ch1, ch2) center predelay level (ch1, ch2) center predelay phase (ch1, ch2)	1 – 30719words 0 – 100% normal/inverse
early reflection time (ch1, ch2) early reflection level (ch1, ch2) early reflection phase (ch1, ch2)	1 – 30719words 0 – 100% normal/inverse
presence control	0.003 – 1.000
spread	0.5 – 2.5
size	0.5 – 2.5

Note:
The smaller the value for the size parameter is, the smaller the maximum effective value for the gate time parameter is.

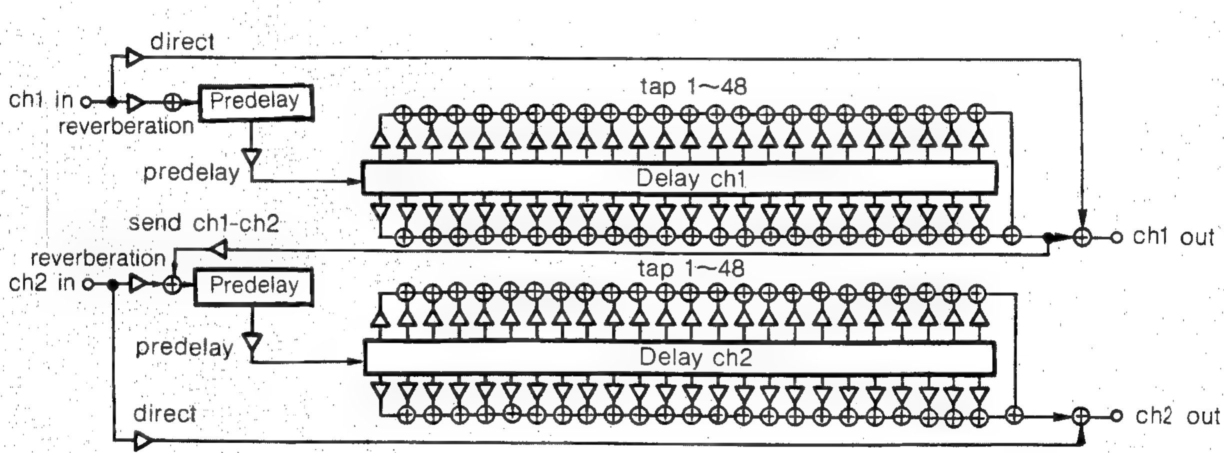


Algorithm 5 Early Reflection (ST-ST) ERF

This is the early reflection of ST-ST type.
You can use the scale or predelay parameter to manipulate the preset which contains many taps.

Parameters	MIN and MAX
predelay time (ch1, ch2) predelay level (ch1, ch2) predelay phase (ch1, ch2)	1 – 32767words 0 – 100% normal/inverse
early reflection1 – 48time (ch1, ch2) early reflection1 – 48level (ch1, ch2) early reflection1 – 48phase (ch1, ch2)	1 – 32767words 0 – 100% normal/inverse

Parameters	MIN and MAX
send ch1 – ch2 level send ch1 – ch2 phase	0 – 100% normal/inverse



Reverberation Block (3) — REV1 and REV2 Blocks

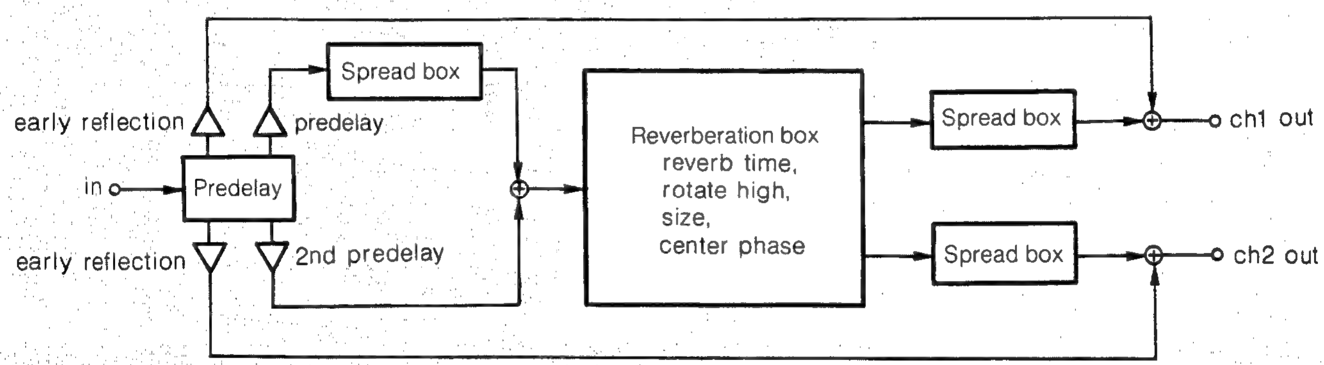
These blocks are used only with MONO-ST type algorithm.

Algorithm 6 Plate Reverberation (MONO-ST) PLR

This is reverberation of MONO-ST type.

Parameters	MIN and MAX
reverb time	0.3 – 99.0sec
predelay time predelay level predelay phase	1 – 19199words 0 – 100% normal/inverse
2nd predelay time 2nd predelay level 2nd predelay phase	1 – 19199words 0 – 100% normal/inverse

Parameters	MIN and MAX
early reflection time (ch1, ch2) early reflection level (ch1, ch2) early reflection phase (ch1, ch2)	1 – 19199words 0 – 100% normal/inverse
rotate high	0.004 – 1.000
spread	0.5 – 1.5
size	0.5 – 1.5
center phase	normal/inverse



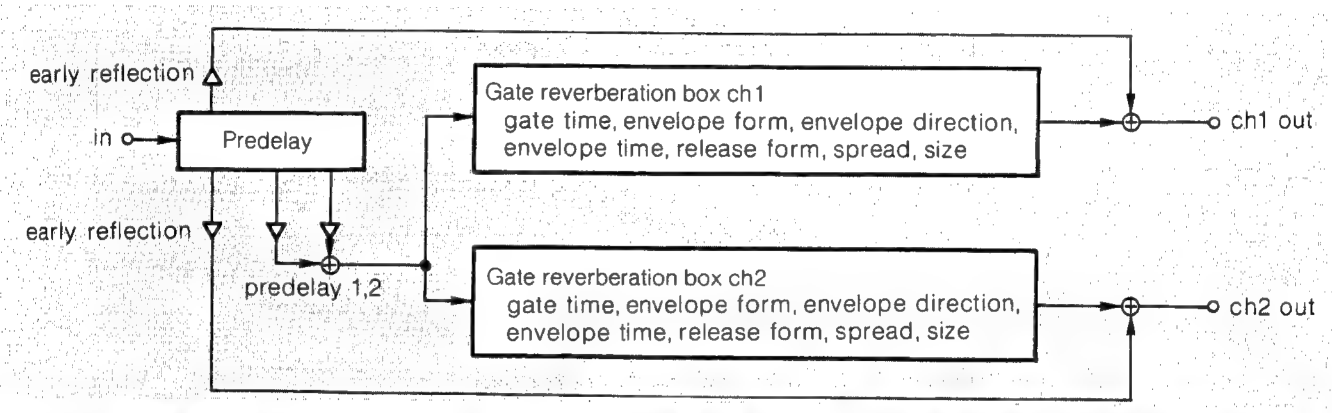
Algorithm 7 Gate Reverberation (MONO-ST) GTR

This is gate reverberation of MONO-ST type.

Parameters	MIN and MAX
gate time (ch1, ch2)	1 – 20480word
envelope form (ch1, ch2) envelope direction (ch1, ch2) envelope time (ch1, ch2)	linear1, linear2 exponential1 exponential2 normal/inverse 0.01 – 99.99sec
release form	5 – 100
predelay1,2 time predelay1,2 level predelay1,2 form	1 – 32767words 0 – 100% normal/inverse

Parameters	MIN and MAX
early reflection time (ch1, ch2) early reflection level (ch1, ch2) early reflection phase (ch1, ch2)	1 – 32767words 0 – 100% normal/inverse
spread	0.5 – 2.5
size	0.5 – 2.5

Note:
The smaller the value for size parameter is, the smaller the maximum effective value for gate time parameter is.

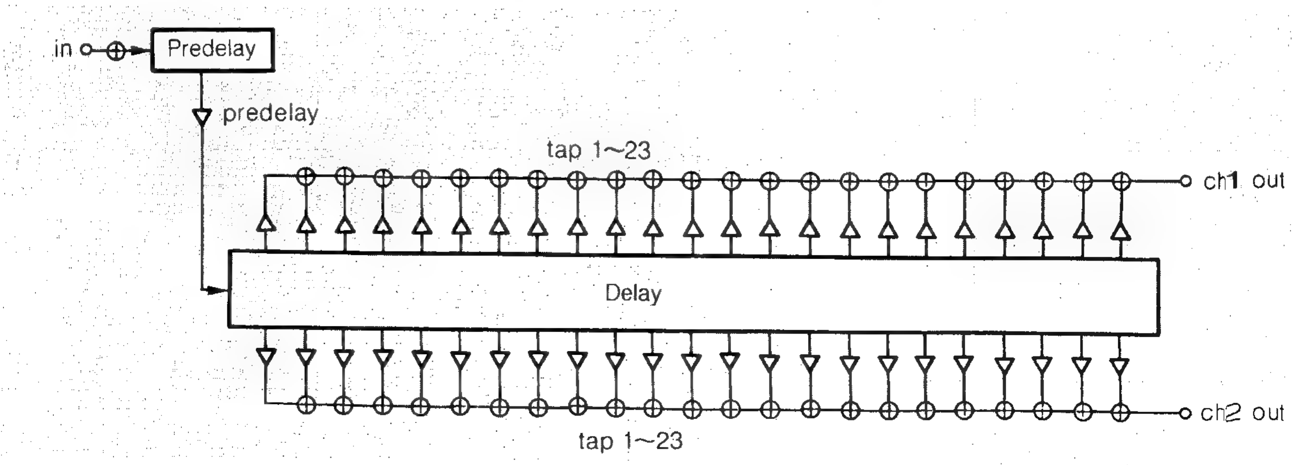


Algorithm 8 Early Reflection (MONO-ST) ERF

This is early reflection of MONO-ST type.

Parameters	MIN and MAX
predelay time predelay level predelay phase	1 – 32767words 0 – 100% normal/inverse

Parameters	MIN and MAX
early reflection1 – 23 time (ch1, ch2) early reflection1 – 23 level (ch1, ch2) early reflection1 – 23 phase (ch1, ch2)	1 – 32767words 0 – 100% normal/inverse



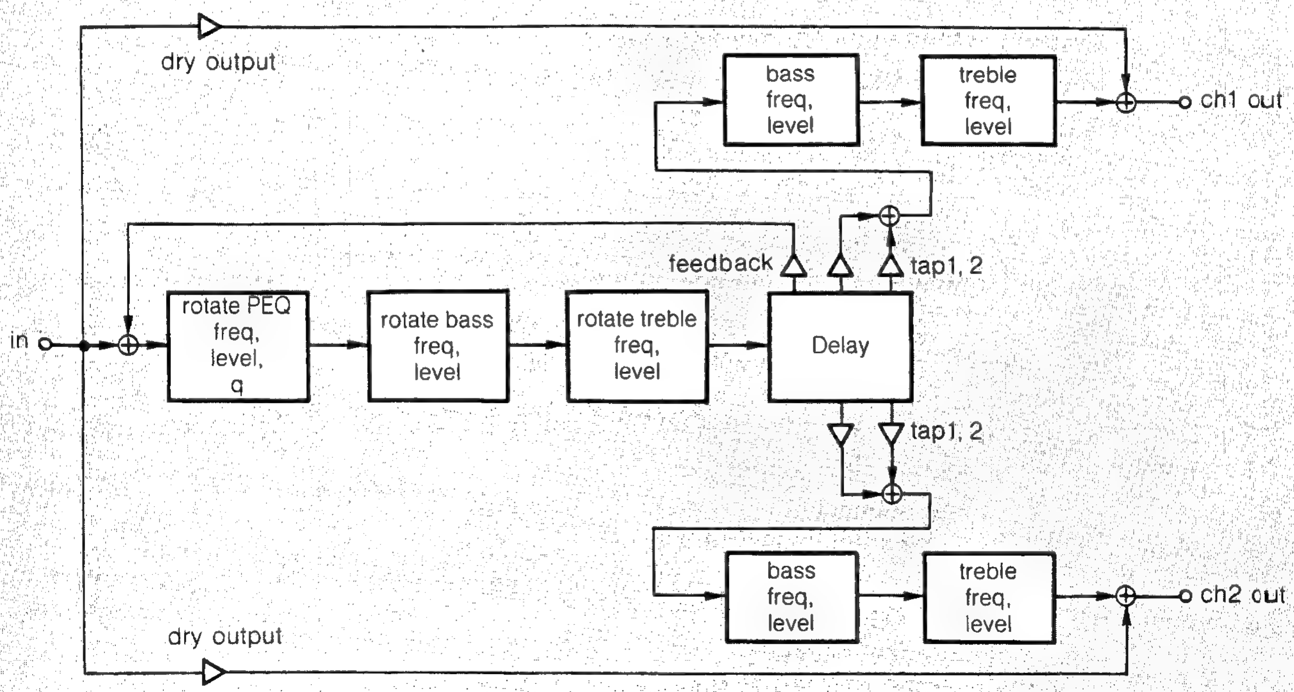
To be continued ►

Algorithm 9 Delay 1 (MONO-ST) DL1

This creates a long delay of MONO-ST type and is equipped with a rotation equalizer and a stereo output equalizer.

Parameters	MIN and MAX
tap1,2 time (ch1, ch2) tap1,2 level (ch1, ch2) tap1,2 phase (ch1, ch2)	1 – 65516words 1 – 100% normal/inverse
feedback time feedback level feedback phase	1 – 65516words 0 – 100% normal/inverse
rotate PEQ freq rotate PEQ level rotate PEQ q	200Hz – 18kHz −12 – +6dB 0.267 – 17.31
rotate bass freq rotate bass level	25Hz – 6.3kHz −12 – +6dB

Parameters	MIN and MAX
rotate treble freq rotate treble level	100Hz – 18kHz −12 – +6dB
bass freq (ch1, ch2) bass level (ch1, ch2)	25Hz – 6.3kHz −12 – +6dB
treble freq (ch1, ch2) treble level (ch1, ch2)	100Hz – 18kHz −12 – +6dB
dry output level (ch1, ch2) dry output phase (ch1, ch2)	0 – 100% normal/inverse

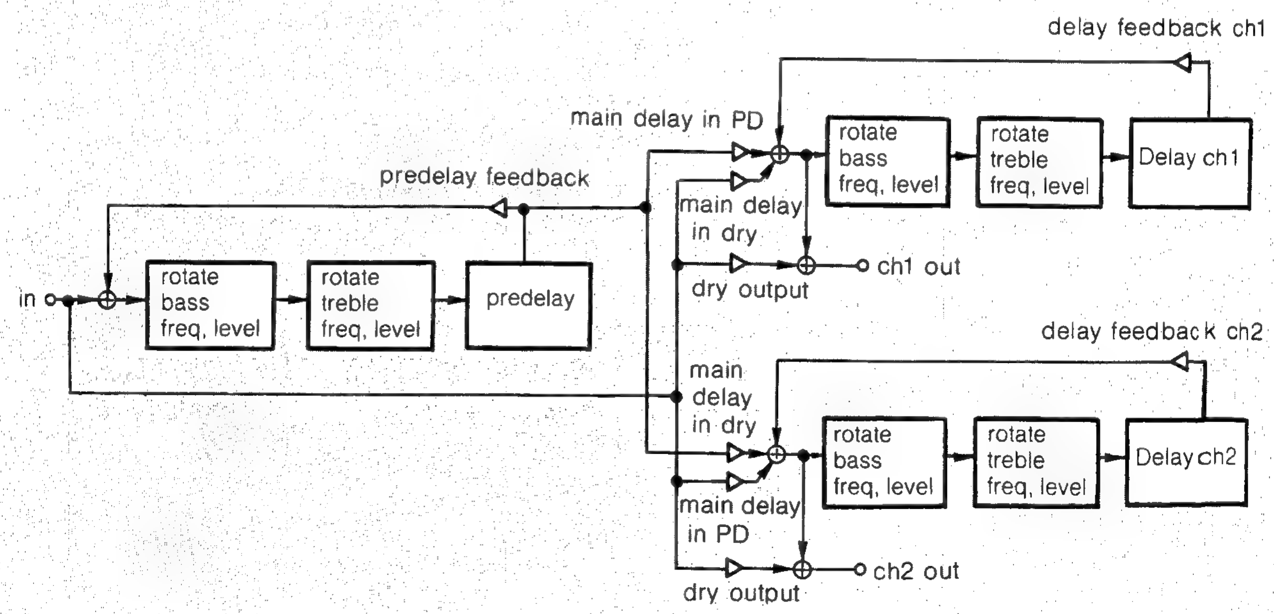


Algorithm 10 Delay 2 (MONO-ST) DL2

This ia a delay combining three types of delays.

Parameters	MIN and MAX
predelay time predelay level predelay phase	1 – 16375words 0 – 100% normal/inverse
predelay feedback level predelay feedback phase	0 – 100% normal/inverse
predelay rotate bass freq predelay rotate bass level	25Hz – 6.3kHz −12 – +6dB
predelay rotate treble freq predelay rotate treble level	100Hz – 18kHz −12 – +6dB
main delay in PD level (ch1, ch2) main delay in PD phase (ch1, ch2)	0 – 100% normal/inverse
main delay in dry level (ch1, ch2) main delay in dry phase (ch1, ch2)	0 – 100% normal/inverse

Parameters	MIN and MAX
delay feedback time (ch1, ch2)	1 – 16379words
delay feedback level (ch1, ch2) delay feedback phase (ch1, ch2)	0 – 100% normal/inverse
delay rotate bass freq (ch1, ch2) delay rotate bass level (ch1, ch2)	25Hz – 6.3kHz −12 – +6dB
delay rotate treble freq (ch1, ch2) delay rotate treble level (ch1, ch2)	100Hz – 18kHz −12 – +6dB
dry output level (ch1, ch2) dry output phase (ch1, ch2)	0 – 100% normal/inverse



Post-effect Block

To be continued ►

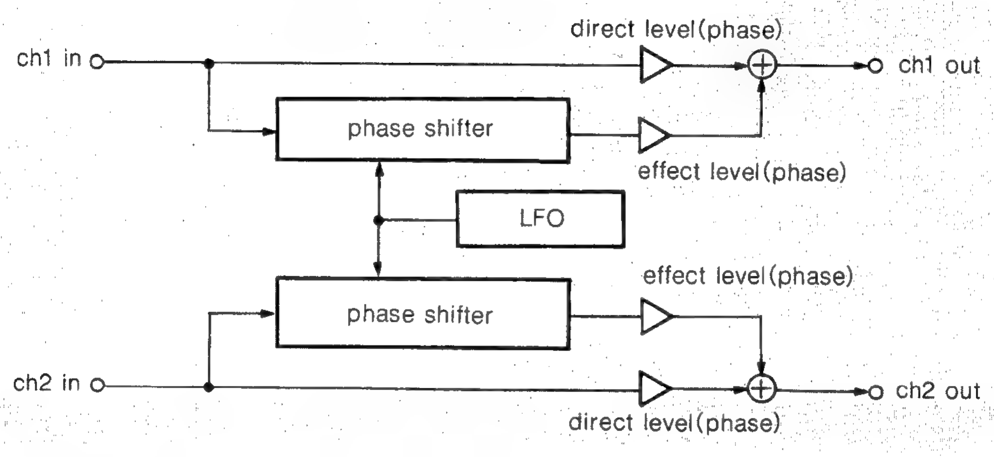
The Post-effect block processes the signals coming from the Reverberation block and then sends them to the Output block. Signal processing in this block uses seven types of specific algorithms for the preset memories.
When editing a preset memory, examine which algorithm is used in the preset memory.

Parameters also vary as the algorithm changes.

Some types of algorithms selectable in the Post-effect block are the same as those in the Pre-effect block. However, be sure that their parameters differ between the two blocks.

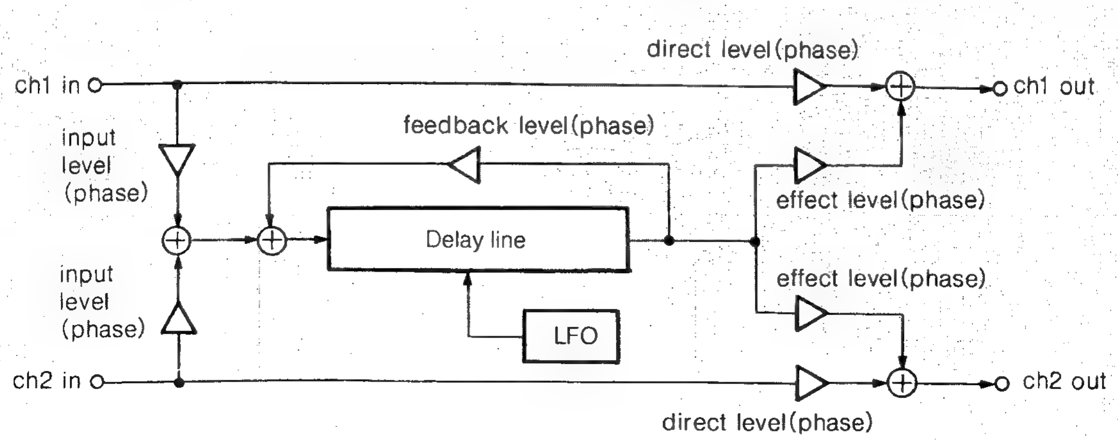
Algorithm 1 Phase Shifter PHS

See page 14 for the parameters.



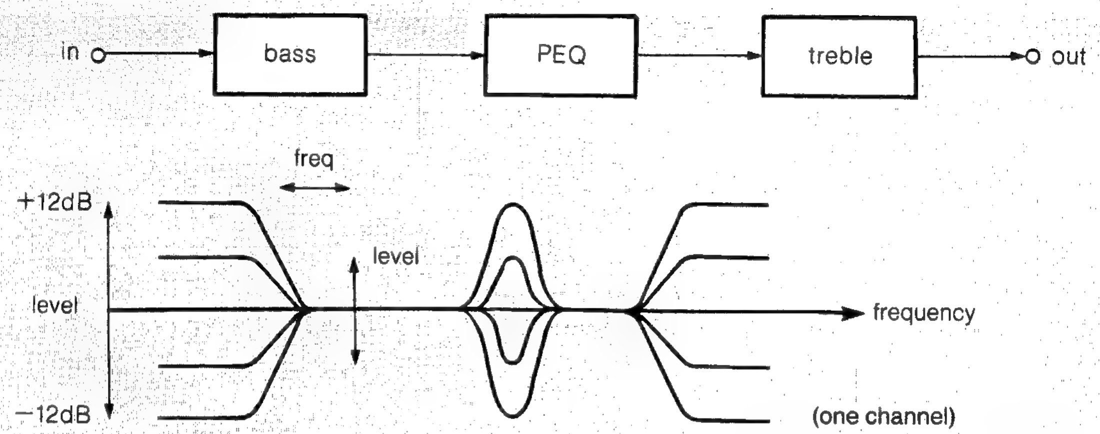
Algorithm 2 Flanger FLG

See page 15 for the parameters.



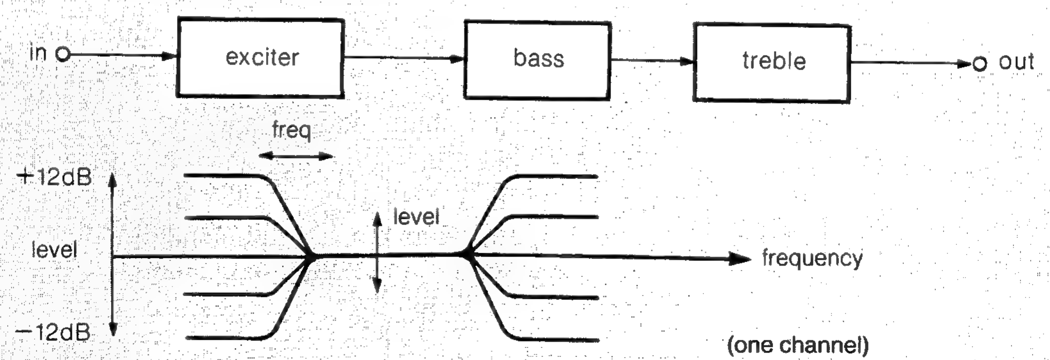
Algorithm 3 Stereo Equalizer SEQ

See page 16 for the parameters.



Algorithm 4 Stereo Exciter + Stereo EQ (Equalizer) SXE

See page 17 for the parameters.

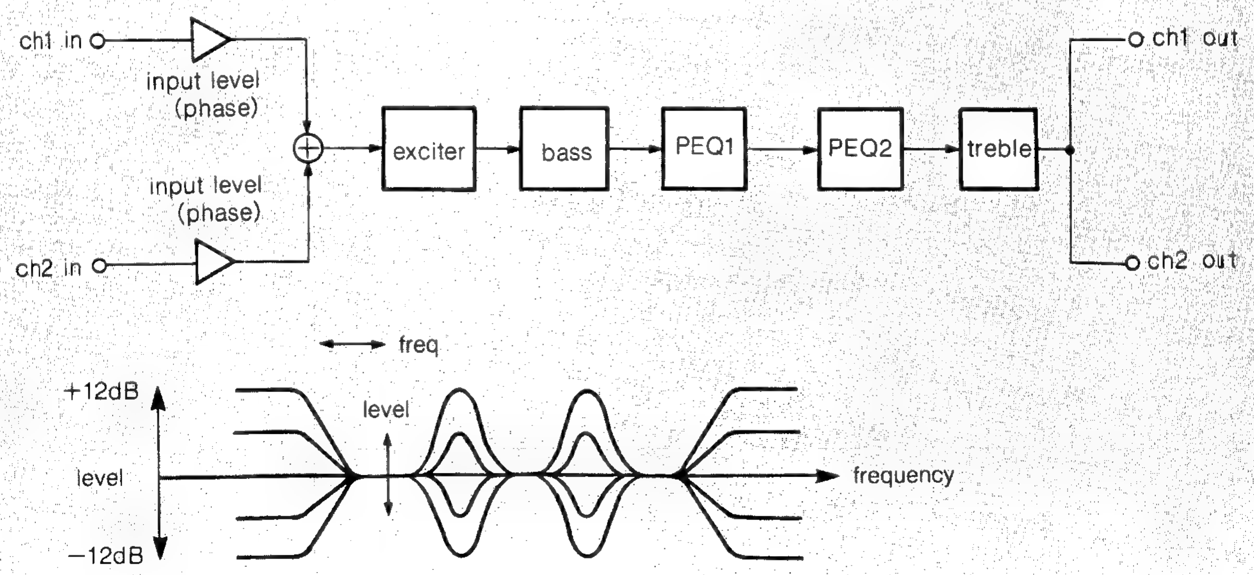


To be continued ►

Algorithm 5 — Monaural Exciter + Monaural EQ (Equalizer) — MXE

This is a monaural digital equalizer with an exciter.
The equalizer consists of 4 band equalization (bass, treble, peaking 1 and peaking 2.)

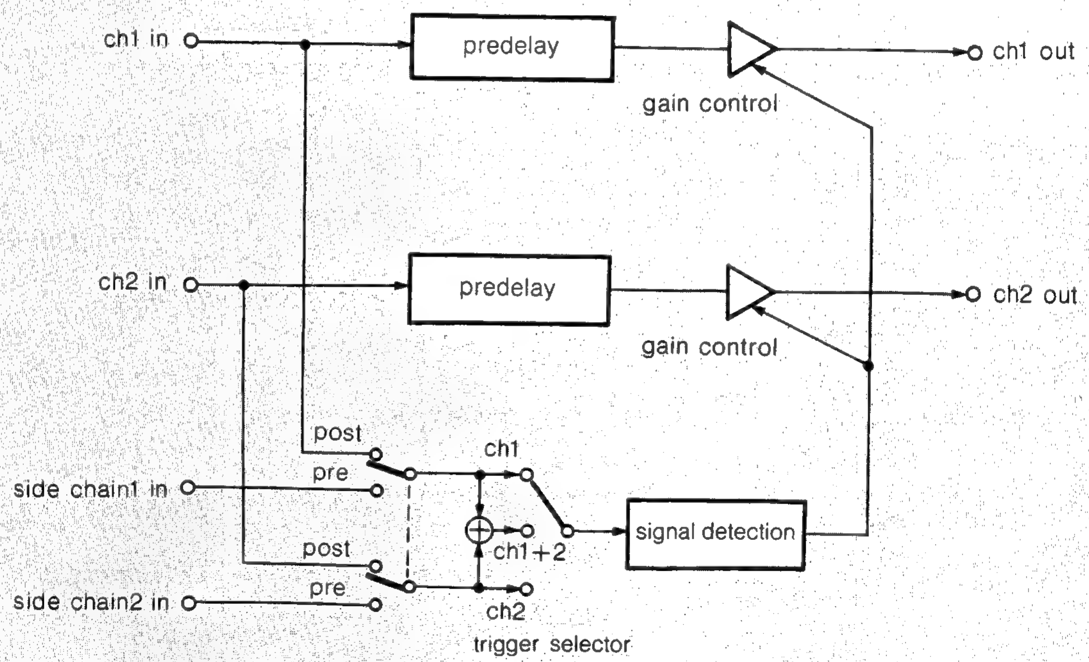
Parameters	MIN and MAX
mono exciter + EQ on/off	on/off
input level (ch1, ch2)	0 – 100%
input phase (ch1, ch2)	normal/inverse
exciter level	−100 – +100%
bass freq bass level	16 – 6.3kHz −12 – +12dB
treble freq treble level	400 – 18.0kHz −12 – +12dB
PEQ1 freq PEQ1 level PEQ1 q	63 – 18.0kHz −12 – +12dB 0.267/0.667/1.414/2.145 4.319/8.651/17.31/34.62
PEQ2 freq PEQ2 level PEQ2 q	63 – 18.0kHz −12 – +12dB 0.267/0.667/1.414/2.145 4.319/8.651/17.31/34.62



Algorithm 6 Gate GTE

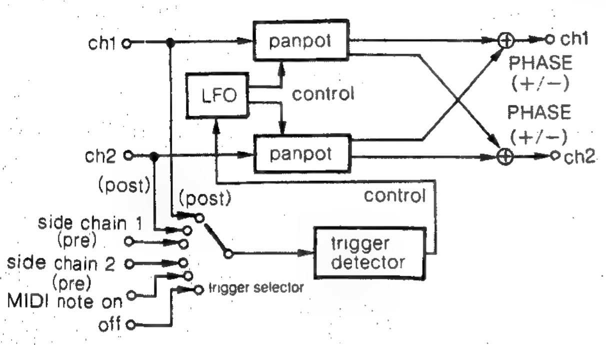
This has the same parameters as the Gate algorithm in the Pre-effect block except that the side chain selecting function is added.
If you select "pre", signals from the Pre-effect block are sent to the signal detecting section through the side chain, whereas if you select "post", signals from the Reverberation block are sent to the signal detecting section.
For other parameters refer to the Gate algorithm in the Pre-effect block (see page 19, Algorithm 6.)

Parameters	MIN and MAX
gate on/off	on/off
trigger select	pre ch1/pre ch2/pre ch1+ch2 post ch1/post ch2/post ch1+ch2
attack time	0 – 500msec
release time	0 – 5000msec
threshold level	0 – 100%
hysterisis level	0 – 100%
predelay time (ch1, ch2)	0 – 10words

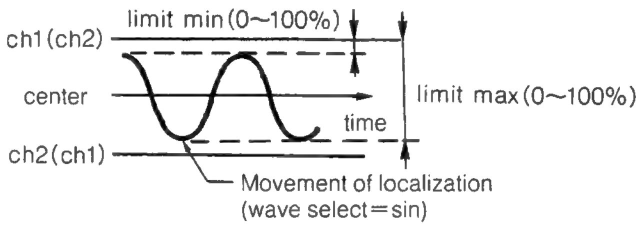


Algorithm 7 Auto Panner APN

This receives the signals from the Reverberation block to give them auto panner processing in stereo.



Parameters	MIN and MAX
autopan on/off	on/off
LFO freq	0.1 – 20Hz
limit min	0 – 100%
limit max	0 – 100%
wave select	sin, triangle, special1, special2
phase (ch1, ch2)	normal/inverse
trigger select	off/pre ch1/pre ch2 post ch1/post ch2/ MIDI note on
trigger threshold	0 – 100%
LFO step LFO start point	1 – 360° 0 – 359°

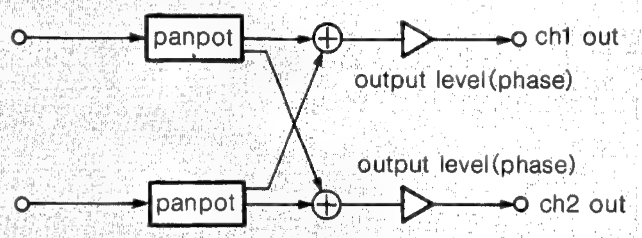
Parameters	Meanings
LFO freq	Specify the LFO frequency.
limit min, of limit max	 This parameter determines the range of the movement of localization as shown in the figure to the left.
wave select	Select the waveform of LFO (Low Frequency Oscillator) that moves the localization.
phase	Set the phase of signals when they are output and mixed to the opposite channel by the auto panner operation.
trigger select	Select the trigger to start the LFO. "pre ch1", "pre ch2", "post ch1" and "post ch2" stand for the sound volume of each channel. "MIDI note on" means that MIDI note on information is the trigger. When "off", the localization keeps on moving.
trigger threshold	Auto panner starts when trigger select is set to "pre ch1", "pre ch2", "post ch1" or "post ch2" and when the signal which is higher than a level set by the trigger threshold parameter comes in. When set to 100%, the trigger will not work.
LFO step, LFO start point	LFO start point determines the initial status of the localization, which moves by the LFO step every time trigger is input. Available range for an LFO step is from 1° to 360° (The localization returns to the original position.) **Note:** Trigger is not accepted while the localization is moving.

Output Block

This block receives the signals from the Post-effect block to give them panpot, level and phase processing. A level control in this block is useful when you want to correct the level difference of each memory.

Parameters	MIN and MAX
output level(ch1, ch2)	0 – 100%
output phase(ch1, ch2)	normal/inverse
panpot(ch1, ch2)	0 – 100%
panpot limit min(ch1, ch2)	0 – 100%
panpot limit max(ch1, ch2)	0 – 100%

Note:
If the panpot is set to 0%, signals pass through the block and if it is set to 100%, input signals for ch1 are output to ch2 and vice versa. Available values for panpot lie between limit min and max.



Other Blocks

To be continued ▶

LCL. MIDI Block

LCL. (LOCAL) MIDI is used when changing tones in real time using MIDI equipment.
It is not intended for rewriting the data in the preset/user memory, but it temporarily changes only the tones based on the data in memory.

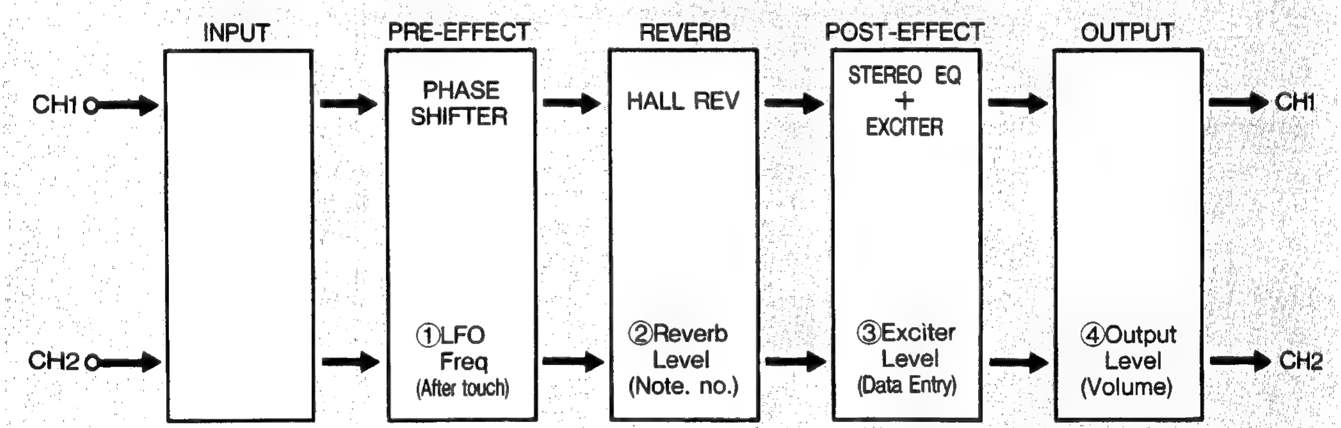
Parameters	Meanings
control no. 1-4	Enter the MIDI control change number. Available values are off, 0-120 or Key velocity, Channel key pressure and Note number.
parameter block 1-4	Select the signal processing block controlled by the number you enter in "control no.1-4".
parameter name 1-4	Select the parameter controlled by the number you enter in "control no.1-4".

Applications for LCL. MIDI

LCL. MIDI is a pioneering function that allows real time control of the internal parameters by means of external MIDI information such as control change number 0 – 120, key velocity, channel key pressure and note numbers, etc.
Moreover it enables simultaneous setting of up to 4 parameters on a memory basis. All the internal parameters besides "time scale" are controllable.

<Sample Application>

The DPS-R7 is connected with a synthesizer.



① Channel Pressure (After touch) controls the LFO frequency of the phase shifter.

② Note number controls the reverberation level. Playing on a higher key enables larger volume of the reverberating sound.

③ Data Entry (controller no.6) controls the exciter level. The data entry volume on the synthesizer performs fine adjustment of the exciter level.

④ Volume (controller no.7) controls output level. A foot volume connected to the synthesizer controls the effect volume directly.

System Block

This block specifies the operating mode for the DPS-R7.

Parameters	Meanings
input mode	Select the input mode (stereo/mono). In monaural mode, only the INPUT CH1 terminals are available.
auto help	Choose to display HELP messages automatically.
load from	Select auto load or enter load. auto load – a memory is automatically called up when you dial the memory number in load mode. enter load – a memory is not called up until you dial the memory number in load mode and then press the ENTER button.
load time	Specify the time which the unit takes to actually indicate a memory called up in the display after the memory number is dialed in auto load mode. The available range is from 200 msec. to 1000 msec.
unit(time)	Specify the unit for time information such as Early Reflection Time and Predelay Time. The available units are word/msec/m/♩/feet(inch). • m indication adopts the calculation in which 1 sec is equal to 340 m • word represents the number of samples.
unit(level)	Specify the unit for level information from %/dB.
unit(q)	Specify the unit for q of EQ. Available units are q/oct.
remote ch	Specify the remote channel from 1 to 15 ch.
remote baud rate	Specify the baud rate of remote control from 9600 to 31250bps.
clock set	Set the calendar and clock. You can move the cursor with the EDIT button. When the parameter menu is displayed, you can examine the clock.
user's name	Enter your name. You can move the cursor with the EDIT button.
date of birth	Enter your birthday. You can move the cursor with the EDIT button.
key protect	This function inhibits any operation even if operation buttons are pressed. This is to prevent any other person from operating the unit by mistake. To release the "key protect", press both the EDIT button and the ENTER button at the same time, then turn the dial counterclockwise.
battery check	Check the battery necessary for maintaning the user memory.
self check	You can verify the software version.

To be continued ►

Memory Block

This block edits the user memory.

Parameters	Meanings
memory compare	For comparative listening with the original memory. The following selections are available. • edit/memory • edit/parameter • edit/parameter/memory • edit/parameter/block/memory edit : Normal editing mode parameter : Only the currently displayed parameter has a value before changing. block : Only the currently displayed block is original. memory : Original data
protect	Turn on/off the memory protection for a designated user memory.
move	Move a user memory to a different number.
copy	Copy a designated preset memory and/or user memory to a different number.
delete	Delete a designated user memory.
exchange	Exchange the two specified user memories.
remaining area	Displays the remaining capacity of the user memory.

SYS.MIDI Block

This block specifies the MIDI operating mode for the DPS-R7.

Parameters	Meanings
MIDI on/off	Turn on for receiving the MIDI data (except for system exclusive messages) (This turns on when tuning on the power.)
OMNI	Set OMNI on/off of MIDI. When OMNI is set to "on", MIDI data is received disregarding the MIDI channel setting.
MIDI ch	Select the MIDI channel from 1 to 16 ch.
bulk dump transfer	Transfer the memory data system information through MIDI. The following information can be transferred. • all(all the user memory, system information and MIDI information). • all user's memory (all the user memories). • system (all the system information). • all MIDI(all the MIDI information). • user's memory (one designated user memory).
bulk dump receive on/off	Turn on for receiving bulk dump. (This turns off when turning on the power.)
program change no.1 – 128	Assign the MIDI program change numbers (1 to 128) to the memory numbers (P1 to P100, U1 to U100, and BYPASS).

B.LOAD (Block load) Block

This block allows you to load the signal processing blocks partially from another memory (preset or user) to the preset memory or user memory being edited. This function corresponds to the Reverberation, Pre-effect, and Post-effect blocks.

Parameters	Meanings
PRE-EF block load	Load the Pre-effect block of another memory (preset or user)
POST-EF block load	Load the Post-effect block of another memory (preset or user)
all REVERB blocks load	Load all the Reveberation blocks (REVC, REVS, REV1 and REV2) of another memory (preset or user).
REVC block load	Load the REVC block of another memory (preset or user).
REVS block load	Load the REVS block of another memory (preset or user). (When ST-ST algorithm is used.)
REV1 block load	Load the REV1 block or the REV2 block of another memory (preset or user) to the REV1 block being edited. (When MONO-ST algorithm is used.)
REV2 block load	Load the REV1 block or the REV2 block of another memory (preset or user) to the REV2 block being edited. (When MONO-ST algorithm is used.)

Sample Indication of B.LOAD

For "PRE-EF (Pre-effect) block load"

```
B.LOAD   ▶PRE-EF block load
[1/6]     U100:PHS [Concert Hall 1  ]
```

Preset memory number or the user memory number

Name of the preset memory or of the user memory (first 16 characters)

Algorithm name of the Pre-effect block

For "all REVERB (Reverberation) blocks load"

```
B.LOAD   ▶all REVERB blocks load
[3/6]     U100:HLR [Concert Hall 1  ]
```

Preset memory number or the user memory number

Name of the preset memory or of the user memory (first 16 characters)

Algorithm name of the reverberation (REVS, REV1 or REV2 block)

For "REVS block load"

REVS appears when you select the ST-ST type algorithm in "all REVERB blocks load".

```
B.LOAD   ▶REVS block load
[5/6]     U100:HLR [Concert Hall 1  ]
```

Preset memory number or the user memory number

Name of the preset memory or of the user memory (first 16 characters)

Algorithm name of the REVS block

For "REV1 block load" and "REV2 block load"

REV1 and REV2 appear when you select the NONO-ST type algorithm in "all REVERB blocks load".

```
B.LOAD   ▶REV1 block load
[5/7]     U256(1):PLR [PLATE + GATE ]
```

```
B.LOAD   ▶REV2 block load
[6/7]     U256(2):GTR [PLATE + GATE ]
```

Preset memory number or the user memory number

(1) is the REV1 block.
(2) is the REV2 block.

Name of the preset memory or of the user memory (first 13 characters)

Algorithm name

B.LOAD block operation

In the B.LOAD block, turning the dial automatically loads the block to the temporary buffer being edited. The time which the unit takes to complete the loading can be set by the "load time" parameter of the System block. Also, you can perform modification and/or comparative listening for parameters of the block that you have newly loaded.

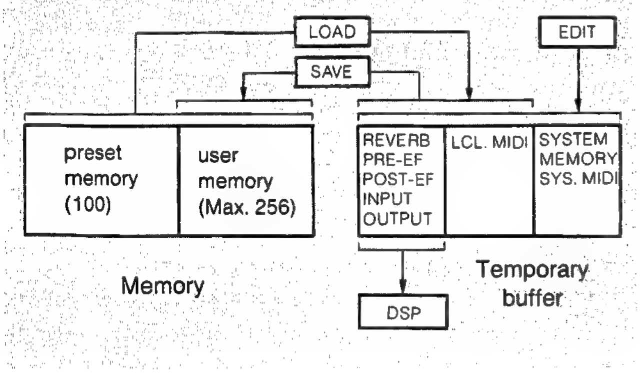
Notes:

- When operating the block load function, the LCL. MIDI parameters are initialized as follows:

control no.	:	off
parameter block	:	INPUT
parameter name	:	input level sync

- Block loading is used, like other parameters, to change the temporary buffer being currently edited. In other words, it is not to directly overwrite the contents of the preset memory or the user memory saved.

Relation between the memory and the temporary buffer

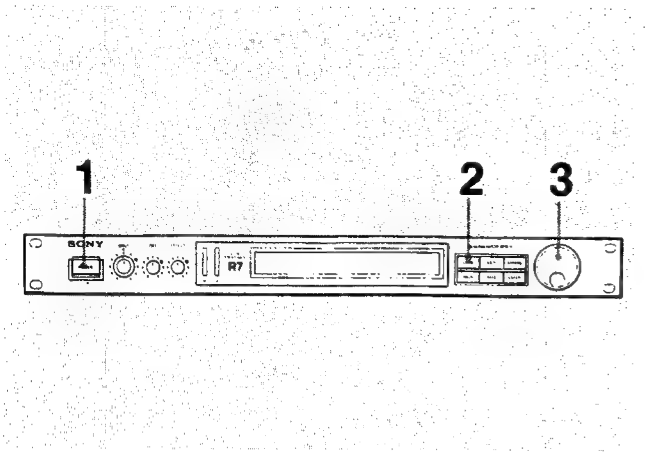


English

Other Blocks

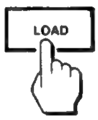
Calling Up a Memory (LOAD)

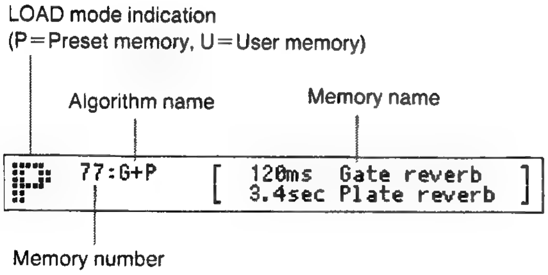
This operation calls up an effect stored in memory.



1. Turn on the power.

2. Press the LOAD button.



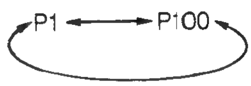


3. Turn the dial and select a desired memory number.

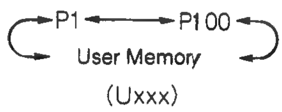
The effect of the selected memory number is automatically called up. When selecting "enter load" for "load from" in the System block, press the ENTER button after selecting the number. (If you select a number different from that of the currently called memory, P or U blinks.)

Memory Numbers
A hundred preset settings are stored in the preset memory in the factory. By turning the dial, these preset settings (P1 to P100) are displayed continuously in order when originally created settings are stored in the user memory, they will be inserted between P100 and P1.

Initial factory setting:



After user memory is registered:



Algorithm names
"Algorithm name" shown in the LOAD mode indication represents the algorithm name of the Reverberation block. The algorithm names for ST-ST type and the MONO-ST type are different from each other in the way they are indicated.

When the algorithm is an ST-ST type :
3 alphabetical characters e.g. HLR, RMR, etc.

When the algorithm is a MONO-ST type :
One character of the REV1 + One character of the REV2.

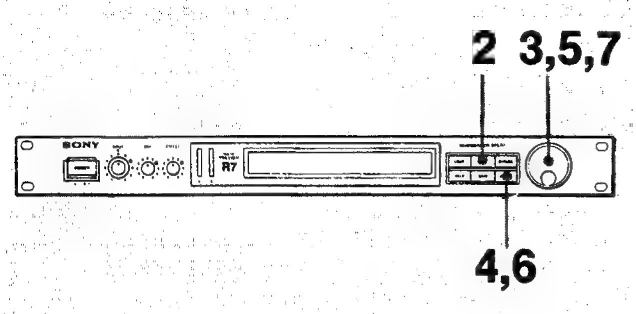
e.g. P+G, E+1
P : PLR(Plate Reverberation)
G : GTR(Gate Reverberation)
E : ERF(Early Reflection)
1 : DL1(Delay1)
2 : DL2(Delay2)

Changing the Effects (EDIT)

To be continued ►

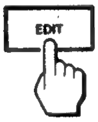
This function allows you to edit the effects saved in memory to create individual ones.

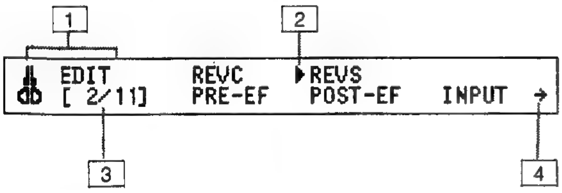
Example : Changing the "reverb time" of the Hall reverberation algorithm



1. Call the memory number to be changed.

2. Press the EDIT button so that the block selecting screen will be displayed.





1 EDIT mode indication

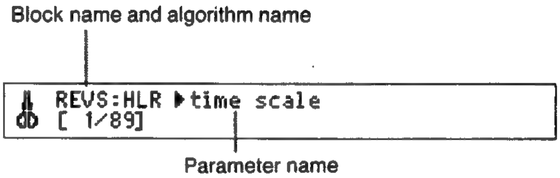
2 Displayed on the left side of the currently selected item.

3 This shows there are 11 selections and the second of these is selected.

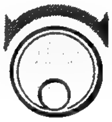
4 This means there are more items following. If not, "↵" appears.

3. Turn the dial and select the block to be changed.

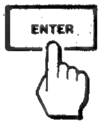
4. Press the ENTER button so that the parameter selecting screen will be displayed.

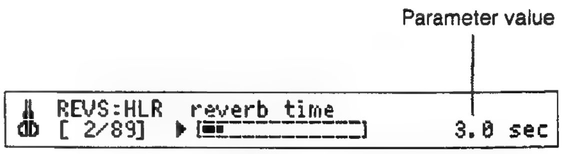


5. Turn the dial and select the parameter to be changed.



6. Press the ENTER button so that the parameter setting screen will be displayed.





The bar graph changes according to the parameter values.

7. Turn the dial and change the parameter value.

To compare the result with the former effect
Press the EDIT button.
This allows you to execute comparative listening in the order you have set with the "memory compare" parameter of the Memory block. By pressing the EDIT button for the second time, the former parameter value will resume.

To change other parameters in the same block
1. After changing a parameter, press the ENTER button.
   The parameter selecting screen will be displayed.
2. Repeat steps 3 to 7 on the previous page to change other parameters as well.

To abort the operation and restore the former memory setting
1. Press the LOAD button.
   Once the former effect resumes, all the parameters you have been setting are deleted, with the message "Parameters have been changed. Are you sure you want to load? Y-ENTER N-EDIT".
   If you accept deletion of the parameters being changed, press the ENTER button. Otherwise, press the EDIT button to store the effect you have created by using the SAVE function. (See page 48.)

2. Press the ENTER button.
   The former memory resumes.

To enter the date and user information in the System block.
Press the EDIT button to move the cursor.

To change other parameters in other blocks
1. After changing a parameter, press the EDIT button.
   The parameter selecting screen will be displayed.

2. Press the EDIT button (or press the ENTER button after selecting QUIT with the dial).
   The block selecting screen will be displayed.

3. Repeat steps 1 and 2 above to do the same for other parameters.

What is time scale?
The time scale is a secondary parameter to change the "delay time" parameters as a whole.
This is set to 100% whenever you select "scale". If at least one of the "delay time" parameters reaches the upper limit by changing the scale, "over" appears. Press the ENTER button after changing this parameter so that each "delay time" parameter changes at the same ratio in its scale. (This parameter will not be stored.)
This is typically used to change the reflection pattern with respect to the size of a room or the tempo of music.

What is sync?
The sync is a secondary parameter for making different parameter values per channel the same. The parameter name is followed by "sync". If ch1, ch2 and center are given different values, executing "sync" forces them to have the same value.

Changing units
Normally the System block is used to select a unit. However, you can also change units while setting a parameter by pressing the ENTER and HELP buttons at once.

Important Points for Editing

1. **Input the actual sound you are going to use.**
   When creating sounds, there is no way besides actually listening to and processing the sounds you actually use. Moreover, do not limit yourself to playing only one instrument to examine the effect, but consider the backup performance (play it during the music) as well, because though you may feel comfortable while listening to the effect played by one instrument, it may not last with the backup performance. Consequently, a strongly applied effect may be found well fitted in the later rendering.

2. **Pay attention to the levels of the DRY and EFFECT controls.**
   Some effects of the preset sounds may greatly change by adjusting the DRY and/or EFFECT control. In particular, those with the message "don't mix with DRY sound" are not effective unless the DRY control is set to 0. When editing, this must be remembered. If you use the Gate for pre/post-effect block, you may fail to get the desired effect with the input below the threshold level. Readjust the input level when this is the case.

3. **Be careful with the tempo.**
   Such effectors as Delay or Reverberator, which are intended for time shift may have diverse grade of effects according to the tempo of music. To solve this problem, you will find that the instruction given in 1 above which says "Input the actual sound and music you are going to use." is critical. That is to say, let the delay time go along with the tempo of music so that you can apply an effect without destroying the rythm of music (Of course there may be cases where you prefer to destroy the rhythm on purpose.)
   The DPS series has a tempo direct indication function that adopts the unit of a quarter note, (how many quarter notes are permitted within one minute) as well as the unit of time (msec) to express the delay time. By gracefully utilizing this function, it becomes easier for you to play music maintaining linkage with the tempo of equipment such as a sequencer.

   * The delay time will be 1 sec. when ♩ = 60 and 500 msec, when ♩ = 120.
   The number you specify should be an integer and a multiple of the tempo.

4. **Use the Time Scaling Function**
   Reverberating function incorporated in the DPS-R7 has many parameters for obtaining a detailed sound creation, and it is also equipped with a time scaling function for speedy editing. It is capable of changing all the parameters categorized in "time scale" at the same time and at the same ratio and greatly helps you edit the time information such as sizes of rooms. Thus, you are allowed to effectively proceed with sound creation as you first call the preset data, organize the rough image of sounds with the time scale function and then change the parameters one by one.

5. **Use the B.LOAD (Block load) Function**
   The B.LOAD function facilitates examination of sounds within one block. Suppose you want to listen to the sounds only in the Reverberation block, you have only to load the OFF algorithm to the Pre/Post-effect blocks. To listen to the original data again, use the memory comparing function on the parameter setting screen of the REVC, REVS(REV 1, REV2), Pre-effect, Post-effect, Input or Output block.

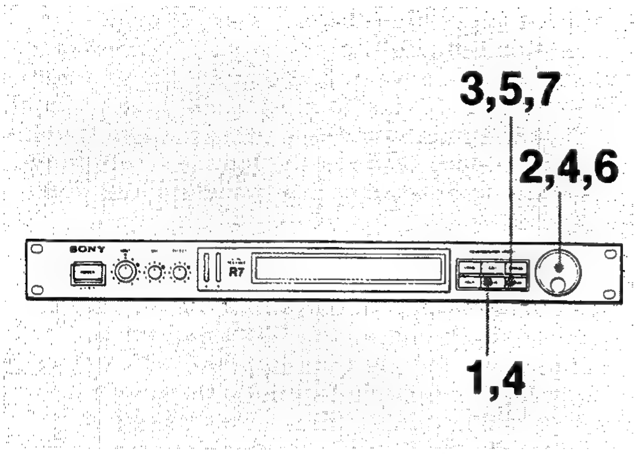
Briefly and finally, what is important in sound creation is as follows:

(1) Select the preset.
(2) Change the block using the B.LOAD function.
(3) Edit with time scaling function.
(4) Edit each parameter.

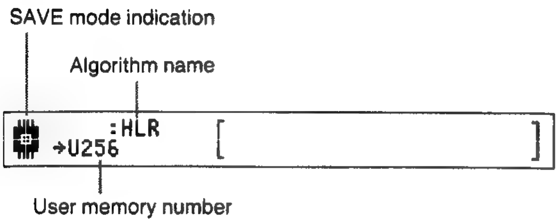
This shows the way of production from the rougher imaging toward detailed editing, which efficiently leads you to the goal.
It should be noted as well, that you minutely compare sounds fully utilizing the comparing function.

Saving the Changed Effects (SAVE)

You can save the changed effects resulting from parameter values you have changed with the Edit function.



1. Press the SAVE button.

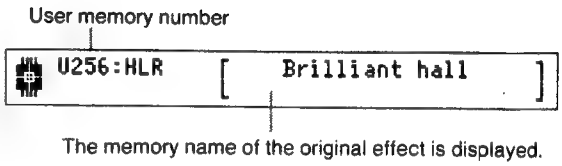


If you designate a memory number which is already stored in memory, you will get the algorithm name and the memory name after the user memory number is displayed.

2. Turn the dial and assign a number to the edited effect.

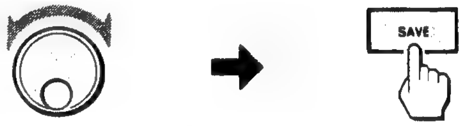
The memory name of the original effect is displayed.

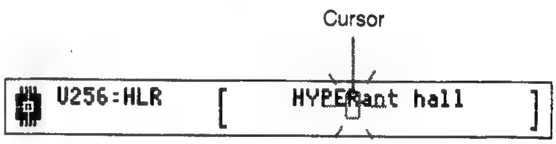
3. Press the ENTER button.



You cannot store the effect in the protected memory number indicated with " " unless you release the protection. (See page 41.)

4. Name the memory.

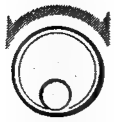




Turn the dial to select characters and press the SAVE button to move the cursor. Everytime the button is pressed, the cursor advances by one character. The characters are lined in the order of 0-9, A-Z, a-z, symbols and dot patterns. To delete the memory name of the original effects, first place the cursor on the heading of memory name indication area. Then turn the dial to select "all clear" and press the ENTER button. ("all clear" lies before the numerals.)

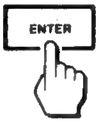
5. Press the ENTER button after naming.

6. Turn the dial and set the memory protection if necessary.



If the memory protection is required, display " " indication

7. Press the ENTER button.



Indication changes to .

When the effect is saved, " " changes to " ". If there is not enough user memory remaining, the effect you have created may not be saved. You will see the message, "Memory full! need xxx byte more.". In this case, you have to delete unnecessary memory (see the Memory block on page 41) so that the remaining memory area will be empty enough to save the effect. If the effect is saved, "completed" is indicated and the SAVE mode is disengaged automatically to the LOAD mode.

What is memory protection?
Memory protection prevents the effect you have saved from being deleted by overwriting. The protected memory number cannot be used for overwriting unless you release the protection in the Memory block.

User Memory List

No.	Name	Algorithm	Description

MIDI Implementation Chart

When using MIDI to transmit data between DPS-R7 and other equipments, it is required to use the following data format below and follow the conditions of the table on page 53.

Channel voice message

```
* 1 1 0 0   n n n n     Program change & channel number(n n n n =0 – 15)
  0 p p p   p p p p     Program number(p p p p p p p =0 – 127)
* 1 0 1 1   n n n n     Control change & channel number(n n n n =0 – 15)
  0 c c c   c c c c     Control number(c c c c c c c =0 – 127)
  0 v v v   v v v v     Control value(v v v v v v v =0 – 127)
* 1 0 0 1   n n n n     Note on & channel number(n n n n =0 – 15)
  0 k k k   k k k k     Note number(k k k k k k k =0 – 127)
  0 v v v   v v v v     Note on velocity(v v v v v v v =0 – 127)
* 1 1 0 1   n n n n     Channel pressure(after-touth) & channel number(n n n n =0 – 15)
  0 v v v   v v v v     Pressure value(v v v v v v v =0 – 127)
```

Channel mode message

```
* 1 0 1 1   n n n n     Mode message & channel number(nnnn=0 – 15)
  0 c c c   c c c c     Omni mode off      Omni mode on
                        ccccccc=124        ccccccc=125
  0 v v v   v v v v     vvvvvvv=0          vvvvvvv=0
```

System exclusive message

```
* 1 1 1 1   0 0 0 0(F0)  Exclusive status
  0 1 0 0   1 1 0 0(4C)  SONY ID                         ⎫
  0 0 0 0   n n n n(0n)  Global channel(nnnn=0 – 15)     ⎬ Exclusive Header
  0 0 0 1   0 0 1 0(12)  DPS-R7 ID                       ⎭
  0 c c c   c c c c      Command
  0 d d d   d d d d      ⎫
        :                ⎬ Data
  0 d d d   d d d d      ⎭
  1 1 1 1   0 1 1 1(F7)  End of Exclusive-EOX
```

ALL DATA DUMP REQUEST (Receive)

```
  Command : 0001 0000(10)
  Data    : None
```

ALL USER MEMORY DUMP REQUEST (Receive)

```
  Command : 0001 0001(11)
  Data    : None
```

SYSTEM DUMP REQUEST (Receive)

```
  Command : 0001 0010(12)
  Data    : None
```

MIDI DUMP REQUEST (Receive)

```
  Command : 0001 0011(13)
  Data    : None
```

USER MEMORY DUMP REQUEST (Receive)

```
  Command : 0001 010n (14or15)
                    bit 7
  Data    : 0  nnn  nnnn              nnnnnnnn : User memory number-1(0 – 255)
            bit 654  3210          bit 76543210
```

ALL DATA DUMP (Send/Receive) (ALL USER MEMORY + SYSTEM + MIDI)

```
  Command : 0001  1000(18)
  Data    : 0ddd  dddd ....        ddddddd : Data(see note 1,6)
```

ALL USER MEMORY DUMP (Send/Receive)

```
  Command : 0001  1001(19)
  Data    : 0ddd  dddd ....        ddddddd : Data(see note 1,2)
```

To be continued ►


```
SYSTEM DUMP (Send/Receive)
  Command : 0001 1010 (1A)
  Data    : 0ddd dddd....          ddddddd : Data (see note 1,2)
MIDI DUMP (Send/Receive)
  Command : 0001 1011 (1B)
  Data    : 0ddd dddd....          ddddddd : Data (see note 1,4)
USER MEMORY DUMP (Send/Receive)
  Command : 0001 110n(1 Cor 1D)
                    bit 7
  Data    : 0 n n n   n n n n
            bit6 5 4   3 2 1 0
          : 0ddd dddd..nnnnnnn       n n n n n n n n  :User memory number-1 (0 – 255)
                                 bit 7 6 5 4 3 2 1 0
                                     ddddddd : Data (see note 1,5)

START ADDRESS TRANSFER (Receive)
  Command : 0010 0000(20)
  Data    : 0 a a a   a a a a
            bit 6 5 4   3 2 1 0
          : 0 a a a   a a a a
            bit D C B   A 9 8 7
          : 0 0 0 a   a a a a        a a a a a a a a a a a a a a a a a a a a  : Start address (0h – 7FFFFh)
            bit    12   1110F E  bit 121110F E D C B A 9 8 7 6 5 4 3 2 1 0

DATA TRANSFER (Receive)
  Command : 0100 0000(40)
  Data    : 0 a a a   a a a a
            bit 6 5 4   3 2 1 0
          : 0 a a a   a a a a
            bit D C B   A 9 8 7
          : 0 0 0 a   a a a a        a a a a a a a a a a a a a a a a a a a : Start address (0h – 7FFFh)
            bit    12   1110F E  bit 121110F E D C B A 9 8 7 6 5 4 3 2 1
          : 0ddd dddd........            ddddddd : (see note1)

note1-dd:Data format
      0ddd dddd   0ddd dddd   0ddd dddd   0ddd dddd–
  bit 765  4321    076 5432    107 6543    210 7654
       ←    dd0    →←    dd1   →←    dd2    →← dd3
      0ddd dddd   0ddd dddd   0ddd dddd   0ddd dddd....
  bit 321  0765    432 1076    543 2107    654 3210
        dd3 →←     dd4    →←   dd5     →←  dd6      →←
note2-ALL USER MEMORY DUMP FORMAT
      dd0   ~ dd513           : USER MEMORY FAT
      dd514 ~ dd121345(max) : USER MEMORY DATA
note3-SYSTEM DUMP FORMAT
      dd0   ~ dd47 : SYSTEM DATA
note4-MIDI DUMP FORMAT
      dd0   ~ dd257 : MIDI DATA
note5-USER MEMORY DUMP FORMAT
      dd0   ~ dd545(max) : USER MEMORY DATA
note6-ALL DATA DUMP FORMAT
      dd0   ~ dd47  : SYSTEM DATA
      dd48  ~ dd305 : MIDI DATA
      dd306 ~ dd819 : USER MEMORY FAT
      dd820 ~ dd121651 : USER MEMORY DATA
```

Universal system exclusive message

INQUIRY MESSAGE

IDENTITY REQUEST (Receive)

* 1 1 1 1	0 0 0 0	(F 0)	Exclusive status	Universal System Exclusive Non-Real Time Header
0 1 1 1	1 1 1 0	(7 E)	Non realtime message	
0 0 0 0	n n n n	(0 n)	Global channel(nnnn=0 – 15)	
0 0 0 0	0 1 1 0	(0 6)	Inquiry message	
0 0 0 0	0 0 0 1	(0 1)	Identity request	
1 1 1 1	0 1 1 1	(F 7)	End of Exclusive – EOX	

IDENTITY REPLY (Send)

* 1 1 1 1	0 0 0 0	(F O)	Exclusive status	Universal System Exclusive Non-Real Time Header
0 1 1 1	1 1 1 0	(7 E)	Non realtime message	
0 0 0 0	n n n n	(0 n)	Global channel(nnnn=0 – 15)	
0 0 0 0	0 1 1 0	(0 6)	Inquiry message	
0 0 0 0	0 0 1 0	(0 2)	Identity reply	
0 1 0 0	1 1 0 0	(4 C)	SONY ID	
0 0 0 0	0 0 0 1	(0 1)		
0 0 0 0	0 0 0 0	(0 0)		
0 0 0 0	0 0 1 0	(0 2)	DPS-R7 ID	
0 0 0 0	0 0 0 0	(0 0)		
0 s s s	s s s s	(s s)	Software version	
0 s s s	s s s s	(s s)		
0 s s s	s s s s	(s s)		
0 s s s	s s s s	(s s)		
1 1 1 1	0 1 1 1	(F 7)	End of Exclusive – EOX	

DIGITAL REVERBERATOR DPS-R7
MIDI Implementation Chart

Date :1. Sep. '91
Version :1.0

Function ...	Transmitted	Recognized	Remarks
Basic Channel — Default Changed	× ×	1 – 16 1 – 16	Memorized
Mode — Default Messages Altered	× × **************	OMNI ON/OFF ×	Memorized
Note Number : True voice	 **************	○ 0 – 127	No sound
Velocity — Note ON Note OFF	× ×	○9n, V=1 – 127 ×	
After Touch — Key's Ch's	× ×	× ○	
Pitch Bend	×	×	
Control Change — 0 – 120	×	○	
Prog Change : True #	× **************	○0 – 127	
System Exclusive	○	○	
Common : Song Pos : Song Sel : Tune	× × ×	× × ×	
System Real Time : Clock : Commands	× ×	× ×	
Aux Messages : Local ON/OFF : All Notes OFF : Active Sense : Reset	× × × ×	× × × ×	
Notes			

Model 1 : OMNI ON, POLY Model 2 : OMNI ON, MONO ○ : Yes
Model 3 : OMNI OFF, POLY Model 4 : OMNI OFF, MONO × : No

Parameter Variation Range for LCL.MIDI Block

The parameters will be changed within the range as shown in the table below when you enter one of the control changes No. 0 – 127.

Parameter variation range for REVC block

Parameters	range
level	0% – Set value
phase	control change data = normal if 0 – 63 = inverse if 64 – 127

Parameter variation range for REVS/REV1/REV2 blocks

parameters	algorithm	range
time	HLR, RMR, PLR, GTR, EFR, P, G, E, 1, 2	Set value–128 words — Set value+126 words
level	HLR, RMR, PLR, GTR, ERF, P, G, E, 1, 2	0% – Set value
phase	HLR, RMR, PLR, GTR, ERF, P, G, E, 1, 2	control change data = normal if 0 – 63 = inverse if 64 – 127
EQ freq	HLR, RMR, 1, 2	Set value–16/6oct. — Set value+15/6oct.
EQ level	HLR,RMR,1,2	0dB – Set value
EQ q	1	0.267 – 17.31
reverb time	HLR, RMR, PLR, P	Set value–64 steps — Set value+63 steps (*1)
spread, size	HLR, RMR, PLR, GTR, P, G	Lower limit — Upper limit (*2)
rotate high	HLR, RMR, PLR, P	Set value–0.25 — Set value+0.25
presence control	HLR, RMR, PLR, GTR	
envelope form	GTR, G	linear1, linear2, exponential1, exponential2
envelope direction	GTR, G	control change data = normal if 0 – 63 = reverse if 64 – 127
envelope time	GTR, G	Set value–1.28 sec — Set value+1.26 sec
release form	GTR, G	5 – 100

To be continued ►

Parameter variation range for pre-effect/post-effect blocks

Parameters	Algorithm	Range
on/off	PHS, FLG, SEQ, SXE, MXE, GTE, APN	control change data = off if 0 – 63 = on if 64 – 127
level	PHS, FLG, SXE, MXE, GTE	0% — Set value
phase	PHS, FLG, MXE (Post-effect)	control change data = normal if 0 – 63 = inverse if 64 – 127
EQ freq	SEQ, SXE, MXE	Set value–16/6oct. — Set value+15/6oct.
EQ level	SEQ, SXE, MXE	0dB — Set value
EQ q	SEQ, SXE, MXE	0.267 – 34.62
LFO freq	PHS, FLG, APN	0.1Hz — Set value
depth	PHS,FLG	Set value — Set value for depth : limit max
depth : limit max	PHS, FLG	depth set value — Set value
attack time	GTE	Set value–128 ms — Set value+126 ms
release time	GTE	Set value–128 steps — Set value+126 steps (*3)
pre delay time	GTE	0 word – 10 words
trigger select	GTE (Pre-effect)	ch1, ch2, ch1+ch2
	GTE (Post-effect)	pre ch1, pre ch2, pre ch1+ch2, post ch1, post ch2, post ch1+ch2
wave select	APN	sin, triangle, special1, special2
autopan phase	APN	control change data = normal if 0 – 63 = inverse if 64 – 127
autopan limit min	APN	Set value — set value for autopan limit max
autopan limit max	APN	Set value for autopan limit min – set value
trigger select	APN	off, pre ch1, pre ch2, post ch1, post ch2, MIDI note on
trigger threshold	APN	0% — set value
LFO step	APN	1° — set value
LFO start point	APN	0° — set value

Parameter Variation Range for LCL.MIDI Block

Parameter variation range for input/output blocks

Parameters	range
level	0% – Set value
phase	control change data = normal if 0 – 63 = inverse if 64 – 127
panpot	0% – Set value
panpot limit min	Set value – set value for panpot limit max (*4)
panpot limit max	Set value for panpot limit min – set value (*5)

If the control by LCL.MIDI exceeds the upper limit of a parameter, the upper limit value is used. On the contrary, the lower limit value is used if it exceeds the lower limit.

(*1) Reverberation time has a different resolution according to the algorithm and the parameter value as shown below. Therefore, available time range differs for the set value even in controlling with LCL. MIDI.

HLR,PLR	0.1sec step if 0.3sec – 10.0sec 1.0sec step if 10.0sec – 99.0sec
RMR	0.04sec step if 0.12 – 4.00sec 0.4sec step if 4.00 – 39.60sec

For example, when the algorithm is HLR and the set value is 10.0sec, the available time range lies between set value−6.4sec and set value+63sec.

(*2) "size" and "spread" have a different range available according to the algorithm used.

(*3) "release time" has a different resolution according to the parameter value as shown below. Therefore available time range differs for the set value even in controlling with LCL. MIDI.

0 – 500 ms	1 ms step
500 ms – 1000 ms	10 ms step
1000 ms – 5000 ms	100 ms step

(*4) In case you change the panpot limit min using LCL. MIDI so that panpot is equal to or smaller than panpot limit min, panpot is also changed to the value of panpot limit min.

(*5) In case you change the panpot limit max using LCL. MIDI so that panpot is equal to or larger than panpot limit max, panpot is also changed to the value of panpot limit max.

Classification Chart for Editing

The chart below lists all the blocks, algorithms and parameters which you can edit. As for the parameters, see reference pages.
The parameters of the Reverberation block and the B.LOAD block varies according to the type of reverberation used.
"ST-ST" below stands for a stereo-in/stereo-out type and "MONO-ST" stands for a monaural-in/ stereo-out type.

"OFF" indicates tells that the reverberation is switched off.
The Reverberation block consists of REVC block and REVS block in ST-ST mode and of REVC block, REV1 block and REV2 block in MONO-ST mode.

BLOCK	ALGORITHM	PARAMETER

- EDIT
  - REVERBERATION (OFF)
    - REVC — see page 58
    - REVS — see page 58
  - REVERBERATION (ST-ST)
    - REVC — see page 58
    - REVS
      - Algorithm1:HLR — see page 58
      - Algorithm2:RMR — see page 58
      - Algorithm3:PLR — see page 58
      - Algorithm4:GTR — see page 59
      - Algorithm5:ERF — see page 59
  - REVERBERATION (MONO-ST)
    - REVC — see page 59
    - REV1/2
      - Algorithm6:PLR — see page 59
      - Algorithm7:GTR — see page 59
      - Algorithm8:ERF — see page 59
      - Algorithm9:DL1 — see page 60
      - Algorithm10:DL2 — see page 60
  - PRE-EFFECT
    - OFF — see page 60
    - Algorithm1:PHS — see page 60
    - Algorithm2:FLG — see page 60
    - Algorithm3:SEQ — see page 60
    - Algorithm4:SXE — see page 60
    - Algorithm5:MXE — see page 60
    - Algorithm6:GTE — see page 61
  - POST-EFFECT
    - OFF — see page 61
    - Algorithm1:PHS — see page 61
    - Algorithm2:FLG — see page 61
    - Algorithm3:SEQ — see page 61
    - Algorithm4:SXE — see page 61
    - Algorithm5:MXE — see page 61
    - Algorithm6:GTE — see page 61
    - Algorithm7:APN — see page 61
  - INPUT — see page 61
  - OUTPUT — see page 61
  - LCL.MIDI — see page 61
  - SYSTEM — see page 61
  - MEMORY — see page 61
  - SYS.MIDI — see page 61
  - B.LOAD
    - (ST-ST) — see page 61
    - (MONO-ST) — see page 61
    - (OFF) — see page 61

Classification Chart for Editing

BLOCK | ALGORITHM | PARAMETER

- EDIT
  - REVC(OFF) — QUIT
  - REVS(OFF) — QUIT
  - REVC (ST-ST)
    - direct level sync/ ch1/ ch2
    - direct phase sync/ ch1/ ch2
    - reverb level sync/ ch1/ ch2
    - reverb phase sync/ ch1/ ch2
    - QUIT
  - REVS (ST-ST)
    - ALG1: HLR
      - time scale
      - reverb time
      - predelay 1 time sync/ ch1/ ch2
      - predelay 1 level sync/ ch1/ ch2
      - predelay 1 phase sync/ ch1/ ch2
      - predelay 2 time sync/ ch1/ ch2
      - predelay 2 level sync/ ch1/ ch2
      - predelay 2 phase sync/ ch1/ ch2
      - cross predelay time sync/ ch1/ ch2
      - cross predelay level sync/ ch1/ ch2
      - cross predelay phase sync/ ch1/ ch2
      - early ref. 1 time sync/ ch1/ ch2
      - early ref. 1 level sync/ ch1/ ch2
      - early ref. 1 phase sync/ ch1/ ch2
      - early ref. 2 time sync/ ch1/ ch2
      - early ref. 2 level sync/ ch1/ ch2
      - early ref. 2 phase sync/ ch1/ ch2
      - 2nd early ref. time sync/ ch1/ ch2
      - 2nd early ref. level sync/ ch1/ ch2
      - 2nd early ref. phase sync/ ch1/ ch2
      - cross early ref. time sync/ ch1/ ch2
      - cross early ref. level sync/ ch1/ ch2
      - cross early ref. phase sync/ ch1/ ch2
      - presence control
      - rotate high
      - rotate bass freq
      - rotate bass level
      - spread
      - size
      - QUIT
    - ALG2: RMR — same as HLR (above)

1 2 3

(to be continued)

BLOCK | ALGORITHM | PARAMETER

1 2 3

- ALG3: PLR
  - time scale
  - reverb time
  - predelay 1 time sync/ ch1/ ch2
  - predelay 1 level sync/ ch1/ ch2
  - predelay 1 phase sync/ ch1/ ch2
  - predelay 2 time sync/ ch1/ ch2
  - predelay 2 level sync/ ch1/ ch2
  - predelay 2 phase sync/ ch1/ ch2
  - 2nd predelay time sync/ ch1/ ch2
  - 2nd predelay level sync/ ch1/ ch2
  - 2nd predelay phase sync/ ch1/ ch2
  - cross predelay time sync/ ch1/ ch2
  - cross predelay level sync/ ch1/ ch2
  - cross predelay phase sync/ ch1/ ch2
  - 2nd cross PD time sync/ ch1/ ch2
  - 2nd cross PD level sync/ ch1/ ch2
  - 2nd cross PD phase sync/ ch1/ ch2
  - early ref. 1 time sync/ ch1/ ch2
  - early ref. 1 level sync/ ch1/ ch2
  - early ref. 1 phase sync/ ch1/ ch2
  - early ref. 2 time sync/ ch1/ ch2
  - early ref. 2 level sync/ ch1/ ch2
  - early ref. 2 phase sync/ ch1/ ch2
  - early ref. 3 time sync/ ch1/ ch2
  - early ref. 3 level sync/ ch1/ ch2
  - early ref. 3 phase sync/ ch1/ ch2
  - 2nd early ref. time sync/ ch1/ ch2
  - 2nd early ref. level sync/ ch1/ ch2
  - 2nd early ref. phase sync/ ch1/ ch2
  - cross early ref. time sync/ ch1/ ch2
  - cross early ref. level sync/ ch1/ ch2
  - cross early ref. phase sync/ ch1/ ch2
  - presence control
  - rotate high
  - spread
  - size
  - QUIT

4 5 6

(to be continued)

To be continued ►

BLOCK	ALGORITHM	PARAMETER

4 5 6

- ALG4: GTR
  - time scale
  - gate time sync/ ch1/ cn2/ center
  - envelope form sync/ ch1/ ch2/ center
  - envelope direction sync/ ch1/ ch2/ center
  - envelope time sync/ ch1/ ch2/ center
  - center output level sync/ ch1/ ch2
  - center output phase sync/ ch1/ ch2
  - release form
  - predelay 1 time sync/ ch1/ ch2
  - predelay 1 level sync/ ch1/ ch2
  - predelay 1 phase sync/ ch1/ ch2
  - predelay 2 time sync/ ch1/ ch2
  - predelay 2 level sync/ ch1/ ch2
  - predelay 2 phase sync/ ch1/ ch2
  - center predelay time sync/ ch1/ ch2
  - center predelay level sync/ ch1/ ch2
  - center predelay phase sync/ ch1/ ch2
  - early ref. time sync/ ch1/ ch2
  - early ref. level sync/ ch1/ ch2
  - early ref. phase sync/ ch1/ ch2
  - presence control
  - spread
  - size
  - QUIT
- ALG5: ERF
  - time scale
  - predelay time sync/ ch1/ ch2
  - predelay level sync/ ch1/ ch2
  - predelay phase sync/ ch1/ ch2
  - send CH1-CH2 level
  - send CH1-CH2 phase
  - early ref. 1 time sync/ ch1/ ch2
  - early ref. 1 level sync/ ch1/ ch2
  - early ref. 1 phase sync/ ch1/ ch2
  - :
  - early ref. 48 time sync/ ch1/ ch2
  - early ref. 48 level sync/ ch1/ ch2
  - early ref. 48 phase sync/ ch1/ ch2
  - QUIT
- REVC (MONO-ST)
  - direct level sync/ ch1/ ch2
  - direct phase sync/ ch1/ ch2
  - reverb level sync/ ch1/ ch2
  - reverb phase sync/ ch1/ ch2
  - send level sync/ ch1/ ch2
  - send phase sync/ ch1/ ch2
  - REV1 output level sync/ ch1/ ch2
  - REV1 output phase sync/ ch1/ ch2
  - QUIT

7 8

(to be continued)

BLOCK	ALGORITHM	PARAMETER

7 8

- REV1 REV2 (MONO-ST)
  - ALG6: PLR
    - time scale
    - reverb time
    - predelay time
    - predelay level
    - predelay phase
    - 2nd predelay time
    - 2nd predelay level
    - 2nd predelay phase
    - early ref. time sync/ ch1/ ch2
    - early ref. level sync/ ch1/ ch2
    - early ref. phase sync/ ch1/ ch2
    - rotate high
    - spread
    - size
    - center phase
    - QUIT
  - ALG7: GTR
    - time scale
    - gate time sync/ ch1/ ch2
    - envelope form sync/ ch1/ ch2
    - envelope direction sync/ ch1/ ch2
    - envelope time sync/ ch1/ ch2
    - release form
    - predelay 1 time
    - predelay 1 level
    - predelay 1 phase
    - predelay 2 time
    - predelay 2 level
    - predelay 2 phase
    - early ref. time sync/ ch1/ ch2
    - early ref. level sync/ ch1/ ch2
    - early ref. phase sync/ ch1/ ch2
    - spread
    - size
    - QUIT
  - ALG8: ERF
    - time scale
    - predelay time
    - predelay level
    - predelay phase
    - early ref. 1 time sync/ ch1/ ch2
    - early ref. 1 level sync/ ch1/ ch2
    - early ref. 1 phase sync/ ch1/ ch2
    - :
    - early ref. 23 time sync/ ch1/ ch2
    - early ref. 23 level sync/ ch1/ ch2
    - early ref. 23 phase sync/ ch1/ ch2
    - QUIT

9 10

(to be continued)

BLOCK	ALGORITHM	PARAMETER

9

10

- ALG9: DL1
  - time scale
  - tap 1 time sync/ ch1/ ch2
  - tap 1 level sync/ ch1/ ch2
  - tap 1 phase sync/ ch1/ ch2
  - tap 2 time sync/ ch1/ ch2
  - tap 2 level sync/ ch1/ ch2
  - tap 2 phase sync/ ch1/ ch2
  - feedback time
  - feedback level
  - feedback phase
  - rotate PEQ freq
  - rotate PEQ level
  - rotate PEQ q
  - rotate bass freq
  - rotate bass level
  - rotate treble freq
  - rotate treble level
  - bass freq sync/ ch1/ ch2
  - bass level sync/ ch1/ ch2
  - treble freq sync/ ch1/ ch2
  - treble level sync/ ch1/ ch2
  - dry output level sync/ ch1/ ch2
  - dry output phase sync/ ch1/ ch2
  - QUIT
- ALG10: DL2
  - time scale
  - predelay time
  - predelay level
  - predelay phase
  - predelay feedback level
  - predelay feedback phase
  - predelay RT bass freq
  - predelay RT bass level
  - predelay RT treble freq
  - predelay RT treble level
  - maindelay in PD level sync/ ch1/ ch2
  - maindelay in PD phase sync/ ch1/ ch2
  - maindelay in dry level sync/ ch1/ ch2
  - maindelay in dry phase sync/ ch1/ ch2
  - delay feedback time sync/ ch1/ ch2
  - delay feedback level sync/ ch1/ ch2
  - delay feedback phase sync/ ch1/ ch2
  - delay RT bass freq sync/ ch1/ ch2
  - delay RT bass level sync/ ch1/ ch2
  - delay RT treble freq sync/ ch1/ ch2
  - delay RT treble level sync/ ch1/ ch2
  - dry output level sync/ ch1/ ch2
  - dry output phase sync/ ch1/ ch2
  - QUIT

11

(to be continued)

BLOCK	ALGORITHM	PARAMETER

11

- PRE-EF
  - OFF
    - QUIT
  - ALG1: PHS
    - phase shifter on/off
    - LFO freq
    - resonance level sync/ ch1/ ch2
    - resonance phase sync/ ch1/ ch2
    - depth
    - depth : limit max
    - direct level sync/ ch1/ ch2
    - direct phase sync/ ch1/ ch2
    - effect level sync/ ch1/ ch2
    - effect phase sync/ ch1/ ch2
    - QUIT
  - ALG2: FLG
    - flanger on/off
    - LFO freq
    - feedback level
    - feedback phase
    - depth
    - depth : limit max
    - input level sync/ ch1/ ch2
    - input phase sync/ ch1/ ch2
    - direct level sync/ ch1/ ch2
    - direct phase sync/ ch1/ ch2
    - effect level sync/ ch1/ ch2
    - effect phase sync/ ch1/ ch2
    - QUIT
  - ALG3: SEQ
    - stereo EQ on/off
    - bass freq sync/ ch1/ ch2
    - bass level sync/ ch1/ch2
    - treble freq sync/ ch1/ch2
    - treble level sync/ ch1/ ch2
    - PEQ freq sync/ ch1/ ch2
    - PEQ level sync/ ch1/ch2
    - PEQ q sync/ ch1/ ch2
    - QUIT
  - ALG4: SXE
    - stereo exciter +EQ on/off
    - exciter level sync/ ch1/ ch2
    - bass freq sync/ ch1/ ch2
    - bass level sync/ ch1/ch2
    - treble freq sync/ ch1/ch2
    - treble level sync/ ch1/ ch2
    - QUIT
  - ALG5: MXE
    - mono exciter +EQ on/off
    - exciter level
    - bass freq
    - bass level
    - treble freq
    - treble level
    - PEQ1 freq
    - PEQ1 level
    - PEQ1 q
    - PEQ2 freq
    - PEQ2 level
    - PEQ2 q
    - QUIT
  - ALG6: GTE
    - gate on/off
    - trigger select
    - attack time
    - release time
    - threshold level
    - hysterisis level
    - predelay time sync/ ch1/ ch2
    - QUIT

12

(to be continued)

BLOCK	ALGORITHM	PARAMETER

12

- POST-EF
  - OFF — QUIT
  - ALG1: PHS — same as in the PRE-EF (Pre-effect) block
  - ALG2: FLG — same as in the PRE-EF (Pre-effect) block
  - ALG3: SEQ — same as in the PRE-EF (Pre-effect) block
  - ALG4: SXE — same as in the PRE-EF (Pre-effect) block
  - ALG5: MXE
    - mono exciter +EQ on/off
    - exciter level
    - bass freq
    - treble freq
    - treble level
    - PEQ1 freq
    - PEQ1 level
    - PEQ1 q
    - PEQ2 freq
    - PEQ2 level
    - PEQ2 q
    - input level sync/ ch1/ ch2
    - input phase sync/ ch1/ ch2
    - QUIT
  - ALG6: GTE — same as in the PRE-EF (Pre-effect) block
  - ALG7: APN
    - autopan on/off
    - LFO freq
    - autopan limit min
    - autopan limit max
    - wave select
    - autopan phase sync/ ch1/ ch2
    - trigger select
    - trigger threshold
    - LFO step
    - LFO start point
    - QUIT
- INPUT
  - input level sync/ ch1/ ch2
  - input phase sync/ ch1/ ch2
  - panpot limit min sync/ ch1/ ch2
  - panpot sync/ ch1/ ch2
  - panpot limit max sync/ ch1/ ch2
  - QUIT
- OUTPUT
  - output level sync/ ch1/ ch2
  - output phase sync/ ch1/ ch2
  - panpot limit min sync/ ch1/ ch2
  - panpot sync/ ch1/ ch2
  - panpot limit max sync/ ch1/ ch2
  - QUIT
- LCL.MIDI
  - control no. 1
  - parameter block 1
  - parameter name 1
  - control no. 2
  - parameter block 2
  - parameter name 2
  - control no. 3
  - parameter block 3
  - parameter name 3
  - control no. 4
  - parameter block 4
  - parameter name 4
  - QUIT

13

(to be continued)

BLOCK	ALGORITHM	PARAMETER

13

- SYSTEM
  - input mode
  - auto help
  - load form
  - load time
  - unit (time)
  - unit (level)
  - unit (q)
  - remote ch
  - remote baud rate
  - clock set
  - user's name
  - date of birth
  - key protect
  - battery check
  - self check
  - QUIT
- MEMOEY
  - memory compare
  - move
  - copy
  - delete
  - exchange
  - remaining area
  - protect U1
  - :
  - protect U256
  - QUIT
- SYS.MIDI
  - MIDI on/off
  - MIDI omni
  - MIDI ch
  - bulk dump transfer
  - bulk dump receive on/off
  - program change no. 1
  - :
  - program change no. 128
  - QUIT
- B.LOAD
  - (ST-ST)
    - PRE-EF block load
    - POST-EF block load
    - all REVERB blocks load
    - REVC block load
    - REVS block load
    - QUIT
  - (MONO-ST)
    - PRE-EF block load
    - POST-EF block load
    - all REVERB blocks load
    - REVC block load
    - REV1 block load
    - REV2 block load
    - QUIT
  - (OFF)
    - PRE-EF block load
    - POST-EF block load
    - all REVERB blocks load
    - QUIT

Specifications

A/D Converter — 18 bit oversampling Stereo A/D converter
D/A Converter — Pulse D/A converter
Sampling Freq. — 40 kHz

Input

Connector type	Reference input level	MAX. input level	Input Impedance	Circuitry type
XLR-3-31 equivalent	+4 dBs	+24 dBs	10 kΩ	Balanced
Phone jack	−10 dBs	+10 dBs	50 kΩ	Unbalanced

XLR-3-31 equivalent connector (1:GND 2:HOT 3:COLD)

Output

Connector type	Reference output level	MAX. output level	Output Impedance	Circuitry type
XLR-3-32 equivalent	+4 dBs	+24 dBs	Min. 600 Ω	Balanced
Phone jack	−10 dBs	+10 dBs	Min. 10 kΩ	Unbalanced

XLR-3-32 equivalent connector (1:GND 2:HOT 3:COLD)

General

Frequency response	10-18 kHz $^{+0}_{-1.0}$ dB
Signal-to-noise ratio	more than 90 dB
Dynamic range	more than 90 dB
Total harmonic distortion	less than 0.004% (1 kHz)
Memory	
Preset memory	100 effects
User memory	maximum of 256 effects
Power requirement	USA and Canadian model 120 V AC, 60 Hz UK model 240 V AC, 50/60 Hz (adjustable with a voltage selector) Continental European model 230 V AC, 50/60 Hz (adjustable with a voltage selector)
Power consumption	28 W
Dimensions	Approx. 482 × 44 × 320 mm (19 × 1 3/4 × 12 5/8 inches) (excluding projections) (w/h/d)
Weight	4.8 kg (10 lb 10 oz)

Design and specifications are subject to change without notice.

Note:
This appliance conforms with EEC Directive 87/308/EEC regarding interference suppression.

Troubleshooting

Symptom	Check if:
Power does not turn on.	• The power cord is not connected to an AC outlet.
No sound is heard.	• The INPUT control is set to 0. • The DRY and EFFECT controls are set to 0.
No sound effect is heard.	• The EFFECT dial is set to 0. • The BYPASS button is pressed.
Sound is distorted.	• Input level is too high. → Turn the INPUT control counterclockwise to lower the level.
No stereo effect is obtained.	• Input mode in the System block is set to "mono".
Uncontrollable with MIDI.	• MIDI receive channel is suitable for send channel of a MIDI equipment. • The control number assigned to this unit is used.

Glossary

Parameter
A factor composing an effect. For example, an initial reflecting sound is composed of factors such as early reflection time and early reflection level, while reverberating sound is composed of pre-delay time and reverberation time. The value of each of these factors is called the parameter and is critical to one solid effect.

Secondary parameter
A parameter capable of modifying parameters while editing under a certain rule, scale and sync are secondary parameters. A secondary parameter is not a real parameter, and cannot be saved, but it can modify more than two parameters at once.

Memory
Internal cicuit board for memorization. To obtain the reverberation effect, the micro computer residing in the unit sends parameters to the signal processing LSI (DSP). You can save these parameters in a memory and call it up when necessary. The memory of the DPS-R7 provides the preset memory for a hundred kinds of effect (preset upon shipment) and user memory (accessible freely by a user) for up to 256 effects.

Temporary buffer
A space where parameters of effects are stored when loading or editing. The parameters stored at this buffer reproduce a corresponding effect.

LOAD
To load means to call up the effect stored in the memory. When loading starts, parameters in the preset memory and/or user memory are copied to the temporary buffer, then new parameters are reflected to digital signal processing.
Partial loading of the memory is controlled by B.LOAD block in the editing mode.

EDIT
To edit means to change the value of parameters. You can create individual effects by changing the parameter values in the temporary buffer.
The EDIT function allows you to improve effects in the preset memory as you desire for the operating condition or your particular needs.

SAVE
To save is to store the parameters in the temporary buffer as part of user memory. This is a significant function for maintaining the individual effects. Once the individual effect is saved, it can be called anytime for editing and/or saving for the second time.

MIDI
MIDI stands for Musical Instrument Digital Interface, a universal standard unit for the data communication among electronic instruments. A keyboard controlling other electronic instruments, a sequencer or a PC organizes automatic playing. The MIDI function incorporated in the DPS-R7 allows you to select the memory number through the MIDI program change number (timbre changing signal for the keyboard), or to control parameters with the MIDI control change signal (variation of modulation wheels).

Algorithm
An operating method required for the digital reverberator to generate a certain effect inside the internal circuit. Each effect of reverberation or gated reverberation is provided with its own operating method. The DPS-R7 incorporates newly developed algorithm enabling various effects far beyond any existing reverberator.

Block Diagram